DOKUZ EYLÜL UNIVERSITY
GRADUATE SCHOOL OF NATURAL AND APPLIED SCIENCES

RELIABLE TRANSPORT FOR WIRELESS SENSOR AND ACTOR NETWORKS

by
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RELIABLE TRANSPORT FOR WIRELESS SENSOR AND ACTOR NETWORKS

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Ph.D. THESIS EXAMINATION RESULT FORM

We have read the thesis entitled **"RELIABLE TRANSPORT FOR WIRELESS SENSOR AND ACTOR NETWORKS"** completed by **FAISAL BASHIR HUSSAIN** under supervision of **ASSOCIATE PROFESSOR YALÇIN ÇEBİ** and we certify that in our opinion it is fully adequate, in scope and in quality, as a thesis for the degree of Doctor of Philosophy.

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RELIABLE TRANSPORT FOR WIRELESS SENSOR AND ACTOR NETWORKS

ABSTRACT

Wireless Sensor and Actor Networks (WSANs) are used for monitoring the physical world, processing data, making decisions and performing appropriate actions. Reliable transport of information in these networks is necessary for the correctness of an appropriate action, for obtaining the exact picture of phenomenon and for updating the modules of sensor nodes.

A scalable, energy-aware and flexible transport solution for WSANs is presented in this study. The proposed transport solution is divided into two major parts sensors-to-actors and actor-to-sensors reliable transport. In order to fulfill different reliability requirements of events, the sensors-to-actors transport is further sub-divided into different transport modes; simple, fair, prioritized and real-time.

Since the sudden impulse of event information from the sensors to the actor results in congestion, a novel congestion control scheme based on packet delivery time and buffer size of nodes is also presented in this study. In order to decrease the affect of interference, a novel schedule based packet forwarding scheme is introduced at the transport layer for orderly delivery of event packets to underlying layers. The actor to sensors reliable transport is aimed to provide successful transport of all data packets from the source to sensor nodes. In this study it is shown that, the rate at which lost packets should be recovered depends on the arrangement of nodes in the network.

Keywords: Sensor networks, transport layer, reliability, rate adjustment, fairness, real-time transport, congestion control.

KABLOSUZ SENSÖR VE AKTÖR AĞLARI İÇİN GÜVENİLİR TAŞIMA

ÖZ

Kablosuz sensör ve aktör ağlar (KSAA), fiziksel dünyayı izlemek, verileri işlemek, kararlar vermek ve uygun eylemlerde bulunmak için kullanılırlar. Bu ağlardaki bilginin güvenilir iletimi, uygun eylemlerin doğruluğu, olguların gerçek görünümü ve sensör düğüm noktalarındaki modüllerin güncellenmeleri için gereklidir.

Bu çalışmada, KSAA için ölçeklenebilir, enerji haberdar ve esnek bir iletim çözümü sunulmaktadır. Önerilen iletim çözümü, sensör-aktör ve aktör-sensör güvenilir iletim olmak üzere iki ana parçaya bölünmüştür. Olayların güvenilirlik gereksinimlerini karşılamak amacıyla sensör-aktör iletim daha sonra dört ayrı iletim moduna ayrılmıştır: basit, adil, önceliklendirilmiş ve gerçek zamanlı.

Sensörlerden aktöre olay bilgilerinin aktarımındaki ani taleplerin tıkanıklığa neden olmasından dolayı, paket iletim süresi ve düğümlerin arabelleklerinin büyüklüğüne dayanan özgün bir tıkanıklık denetim planı da bu çalışmada sunulmuştur. Girişimin etkisini azaltmak amacıyla, iletim katmanında, alttaki katmanlara düzenli olay paketi iletimi için, özgün bir tarife tabanlı paket yönlendirme planı belirtilmiştir. Aktör-Sensör güvenilir iletim, tüm veri paketlerinin kaynaktan tüm sensör düğüm noktalarına tatminkar iletimini sağlamayı amaçlamaktadır. Bu çalışmada, kayıp paketlerin kazanılabildiği iletim oranının, ağdaki düğüm noktalarının düzenlenmesine bağlı olduğu gösterilmiştir.

Anahtar sözcükler: Sensör ağları, taşıma tabanı, güvenlilir, hız ayarlama, eşitlik, zaman bağlı taşıma, sıkışıklık kontrol.

CONTENTS

CHAPTER ONE
INTRODUCTION

1.1 Overview

Wireless Sensor Networks (WSNs) gather information from the environment by measuring mechanical, thermal, biological, chemical, optical, and magnetic phenomena. The electronics then process the information derived from the sensors and through some decision making capability direct critical information to the sink. Sensor network architecture is shown in Figure 1.1 where nodes communicate with a sink (base station) which is capable of communicating with the user (manager node) through Internet or a satellite link.

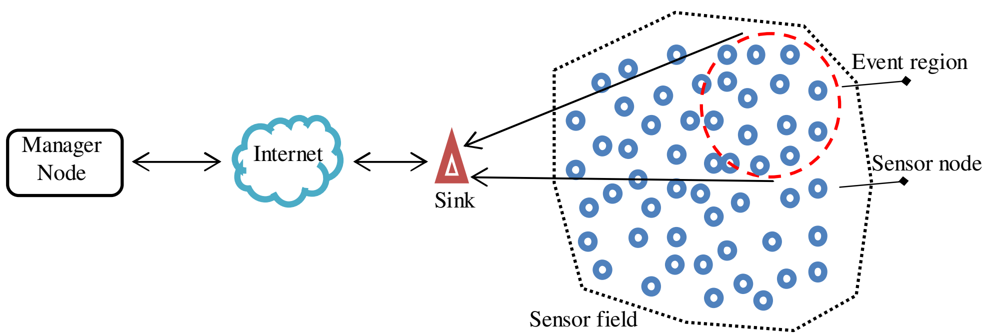

Figure 1.1 Wireless sensor network architecture.

The research in the field of wireless sensor networks has lead to the emergence of Wireless Sensor and Actor Networks (WSANs). Wireless Sensor and Actor Networks have shifted the information gathering phenomenon of wireless sensor networks to the new era of decision making and controlling the environment. Sensor and actor (sometimes referred as actuators) in these networks are capable of observing the physical world, processing data, making decisions and performing appropriate action. The phenomenon of sensing and acting is performed by sensor and actor respectively, in a highly coordinated manner (Akyildiz & Kasimoglu, 2004). Sensors (like in WSNs) are low-cost, low power and limited energy devices which sense external environmental conditions; also termed as *sensor nodes*. Actors also termed as *actor nodes* which are resource rich devices and their basic task is decision making and taking necessary action. A simple wireless sensor and actor network architecture is shown in Figure 1.2 where nodes send event information to closet actor, which take appropriate action and send the information to the sink.

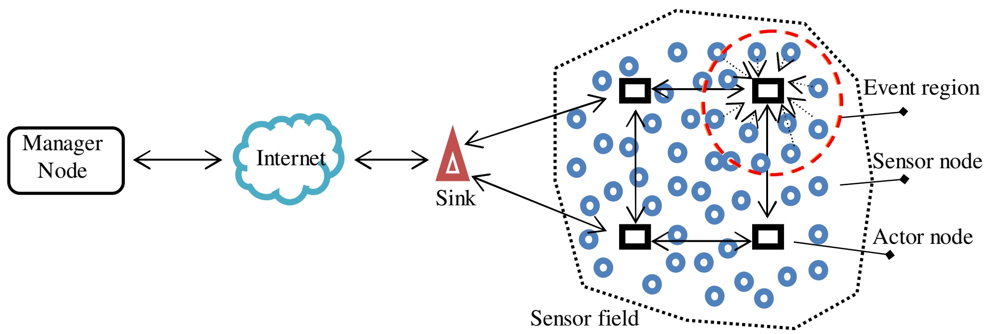

Figure 1.2 Wireless sensor and actor network architecture.

WSNs have gained incredible recognition in the last few years (Chong & Kumar, 2003; Culler, Estrin, & Srivastava, 2004). This is mainly due to the advancements in the Micro-Electro-Mechanical Systems (MEMS) technology (Gardner, Varadan, & Awadelkarim, 2001). Smaller but more efficient sensors in terms of sensing, processing, storage and energy are in use (Warneke, Last, Liebowitz, & Pister, 2001; Hill & Culler, 2002). Sensor networks are standalone networks in which sensor nodes continuously or periodically sense their surrounding environment. These nodes can be placed or scattered at different locations for sensing purpose. Sensor nodes generally send information to a target (sink) upon the occurrence of an event. Therefore, these networks are also termed as *event driven networks* (Akan & Akyildiz, 2005). An *event* is anything of interest for the application e.g., fire, leakage of a poisonous gas, increase in pressure etc. The sensing capability coupled in a small size box with processing and wireless transmission capabilities, allow these networks to be setup in small time with great degree of effectiveness for wide range of applications. Moreover, the introduction of actors to these networks has further enhanced the effectiveness of sensor networks. As a result, sensor networks are now an integral part of systems like battlefield surveillance, microclimate control in buildings, biological and chemical attack detection, smart environments and in different disaster recovery applications (Raghavendra, Sivalingam, & Znati, 2004).

The communication in sensor networks is subject to various physical, operational and environmental limitations. Sheer numbers of inaccessible and unattended sensor nodes, which are prone to frequent failures, make topology maintenance and communication a challenging task (Tilak, Abu-Ghazaleh, & Heinzelman, 2002). Conventional communication protocols for wireless networks are not considered suitable for these networks because they do not take into account the limitation of sensor networks (Akyildiz, Su, Sankarasubramaniam, & Cayirci, 2002). New routing techniques (Al-Karaki & Kamal, 2004; Ganesan, Govindan, Shenker, & Estrin, 2002; Intanagonwiwat, Govindan, Estrin, Heidemann, & Silva, 2002), design of energy efficient MACs (Polastre, Hill, & Culler, 2004; Ye, Heidemann, & Estrin, 2002) and topology control (Cerpa & Estrin, 2002; Chen, Jamieson, Balakrishnan, & Morris, 2002) for conservation of energy in sensor networks has been the focus of most of the researchers in the recent years. However, with the increase in the application areas of wireless sensor and actor networks reliable transport, synchronization and mobility of nodes have gained rapid recognition. The addition of actors to these networks have bring forth some new issues (Akyildiz & Kasimoglu, 2004), because the normal single destination (sink) phenomenon of WSNs is shifted to multiple destinations (actors) in WSANs. As a result, new solutions are subject to research for WSANs.

1.2 Problem Statement

The basic purpose of a transport layer is to provide a mechanism for reliable or guaranteed information transfer between source and destination. Hence, methods for congestion detection, mitigation and rate control are implicitly included in a transport layer. The information which is to be transported in sensor networks is comprised of some readings of sensors, which nodes send to destination upon event occurrence (Akan & Akyildiz, 2005). Also, the information may contain a binary file that a destination sends to sensor nodes for updating their event database or for completely changing the binary code running on sensors (Wan, Campbell, & Krishnamurthy, 2005). Transport of different events information at the same time, fairness and time-bound event transport are some of the important aspects of a transport layer in wireless sensor networks (Chong & Kumar, 2003). Apart from this, prior to information transport by sensors selection of appropriate actor and actor-to-actor coordination for decision making are some transport layer requirements specific to WSANs.

The research in the field of reliable information transport in WSNs and WSANs networks focuses on providing individual solutions for various aspects of transport in sensor networks. The existing solutions have been proposed for different applications, spanning on different layers of protocol stack with contradicting basic assumptions. Therefore, due to the architectural and operational differences of these solutions, existing protocols are not appropriate to operate in a unified manner at the transport layer of sensor networks.

1.3 Thesis Contribution

This work presents a scalable, energy-aware and flexible transport mechanism for wireless sensor and actor networks that is responsible for reliably transporting information from actor-to-sensors, sensors-to-actors with a congestion control scheme. The transport solution provides high degree of reliability with minimum energy consumption for scalable and dense wireless sensor and actor networks. The transport solution presented in this study is independent of underlying routing and medium access layer.

Actor-to-sensors event transport guarantees transfer of information to all the destination nodes even under high channel error conditions. However, sensors-to-actors transport either aims to achieve application defined throughput or provides maximum throughput for reliable event detection. The proposed transport mechanism depending on the nature of event can switch to an appropriate sensors-to-actors transport mode. The sensors-to-actors transport contains following transport modes:

- Simple transport mode for reliable delivery of general event information to actor(s) from the event region.
- Fair transport mode that shares the bandwidth among all the event reporting nodes in order to provide same per node throughput at the actor(s).
- Multiple event transport mode for reliable delivery of multiple events information to actor(s), which are occurring at the same time within the event region.
- Time-bound transport mode for delivering time critical event information to actor(s).

As a summary, this study contributes to the existing research on reliable transport for sensor networks by presenting a unified transport solution which can be used either partly or as a single unit for different applications. According to existing literature review, no transport protocol for WSNs or WSANs has been found that encompasses various transport modes in a single transport solution.

1.4 Thesis Outline

The thesis is organized in the following manner; in chapter two the related work on transport protocols for WSN and WSANs is presented. Refereeing to existing literature the reasons why normal wired and wireless transport protocols are not suitable for sensor networks are also discussed in chapter two. The architecture of wireless sensor and actor networks, along with design challenges for communication protocol in these networks are presented in chapter three. A mining application is discussed in detail to understand different information flows and the necessity for a transport solution. In chapter four, the proposed transport mechanism for WSANs along with basic definitions, assumptions and goals is presented. A congestion control protocol that mitigates congestion during sensors-to-actor transport is presented in chapter five. Detailed simulation results are shown to evaluate the performance of the proposed congestion control protocol. Simple, fair and prioritized event transport modes along with simulation results are presented in chapter six. Two real time event transport schemes are presented in chapter seven. In chapter eight, an actor-to-nodes information transport protocol which provides guaranteed packet delivery is presented. In the last chapter, this study is concluded along with its major achievements and future directions.

CHAPTER TWO

RELATED WORK

The term sensor network has been around for more than a couple of decades (Chong & Kumar, 2003). The large size of sensors, with separate sensing, communication and processing units had kept these networks out of the main stream research (Ilyas & Mahqoub, 2004). In the recent years, advances in miniaturization technology now allows to bundle small sensors, low-power circuits, wireless communication equipment and small-scale energy supplies in one small piece; as the new generation of sensor nodes (Akyildiz & Kasimoglu, 2004; Karl & Willig, 2005).

2.1 Wireless Sensor Networks and Ad Hoc Networks

Wireless sensor networks are distributed systems communicating with each other using radio communication (Culler, Estrin, & Srivastava, 2004; Pottie & Kaiser, 2000). The random deployment of these sensor nodes, standalone state of operation and the use of wireless communication infer that these networks are similar to wireless ad hoc networks. Although there are similarities among ad hoc networks and wireless sensor networks still the ad hoc network's communication protocols (Chlamtac, Conti, & Liu, 2003) are not suitable for sensor networks (Karl & Willig, 2005). To illustrate this point, some of the major differences between sensor networks and ad hoc networks as indicated by Akyildiz & Kasimoglu (2004) and Chong & Kumar (2003) are given as follows.

- The number of sensor nodes in sensor networks can be of several orders of magnitude higher than the nodes in an ad hoc network.
- Sensor nodes are densely deployed.
- Sensor nodes are prone to failures both physical and operational.
- The topology of a sensor network changes very frequently due to loss of sufficient energy or mobile nodes.
- Sensor nodes mainly use broadcast communication paradigm whereas most ad hoc networks are based on point-to-point communications.

- Sensor nodes are limited in power, computational capacities, and memory.
- Sensor nodes may not have global identification (ID) because of the large amount of overhead and large number of sensors.

In wireless sensor networks, large numbers of sensor nodes are densely deployed, so neighbor nodes may be very close to each other. Hence, multi-hop communication in sensor networks is expected to consume less power than the traditional single hop communication (Intanagonwiwat & et al., 2002; Rabaey, Ammer, Dasilva, Patel, & Roundy, 2000). The transmission power levels can be kept low, which is highly desired in covert operations. Multi-hop communication can also effectively overcome some of the signal propagation effects experienced in long-distance wireless communication (Rappapport, 2002; Schwartz, 2004).

One of the most important constraints on sensor nodes is the low power consumption requirement (Karl & Willig, 2005). Sensor nodes carry limited, generally irreplaceable, power sources. Traditional networks aim to achieve high quality of service (QoS) provisions while sensor network protocols must focus primarily on power conservation. They must have inbuilt trade-off mechanisms that give the end user the option of prolonging network lifetime at the cost of lower throughput or higher transmission delay.

As summarized, the communication protocols for ad hoc networks are not suitable for wireless sensor networks (Akyildiz & Kasimoglu, 2004 and Chong & Kumar, 2003). Therefore, for the last few years, significant research has been conducted on the creation of new medium access protocols (Dam & Langendoen, 2003; Shin, Kim, & Hwang, 2007; Ye, Heidemann, & Estrin, 2002) for low-power radio communications. New topology management protocols (Cerpa & Estrin, 2002; Chen & et al., 2002) are suggested to activate as minimum as possible nodes for efficiently monitoring the surrounding environment; with minimum energy consumption. Also, new routing paradigms for clustered and non-cluster network topologies (Heidemann, Silva, & Estrin, 2003; Heinzelman, Kulik, & Balakrishnan, 1999; Shah, Bozyigit, Hussain, & Akan, 2006) have been suggested to provide multi-hop information routing with minimum overhead.

2.2 The Issue of Reliable Transport in Wireless Sensor and Actor Networks

The importance of a reliable transport mechanism in sensor networks has been pointed out by Akyildiz & et al. (2002) and Pottie & Kaiser (2000). According to Akyildiz & Kasimoglu, (2004), reliable transport protocols and congestion control mechanisms for wireless sensor networks have got late recognition from the researchers. Since energy conservation is the basic issue, the introduction of a transport solution increases the energy consumption by making extra reliability related transmissions. Wireless sensor networks are event driven networks and on event occurrence due to dense nature of these networks, a number of nodes detecting the event transmit information to destination(s). Redundant data travelling through multiple flows is forwarded to destination(s) (Akan & Akyildiz, 2005) and occasional loss of information is not deemed to affect the overall information delivery to the destination(s) (Cerpa, Elson, Hamilton, & Zhao, 2000). Hence, the presence of redundant information flows in these networks decreases the need for a transport solution.

Wireless sensor networks are application dependent networks (Cook & Das, 2004), therefore the issue of transport is also application dependent. The shifting of these networks from research labs to industry and the increase in the application areas of WSNs, demands for different reliability standards at the transport layer (Wang & et al. 2005). When a wireless sensor and actor network for forest fire detection and control is considered, this network might not require a high degree of reliability (a transport solution). In this network, in case of fire, multiple sensors send fire information to nearby actors. Therefore, the possibility that some actors receive the fire information is still very high. On the other hand, the application can require that for correct event detection, certain amount of information must reach actors; in order to trigger the water sprinklers. For identifying the exact number and location of water sprinklers that can effectively extinguish fire, precise per node event information is necessary. Hence, the level/degree of reliability is application defined (Akan & Akyildiz, 2005).

Wireless sensor networks are standalone networks in which large numbers of nodes operate in an unattended fashion. Therefore, Wan, Campbell, & Krishnamurthy, (2005) suggests that wireless sensor networks not only require reliable transport but also guaranteed information delivery. For example, in the forest fire detection and control application, if the sensors are required to detect an additional event or it is required to change an event's definition then an update of binary code running on sensor nodes is necessary. These kinds of applications require guaranteed transport of information to all the nodes in the network. Physically locating thousands of small sized sensors randomly scattered in the forest and changing the binary code, might not be possible.

Another issue related to transport layer in WSNs is that of congestion. The importance of congestion control has been indicated in the works of Tilak, Abu-Ghazaleh, & Heinzelman, 2002 and Akyildiz & et al. 2002. In case of event occurrence, the sudden flow of information from event nodes to a single or few destinations results in congestion (Wan, Eisenman, & Campbell, 2003). The degree of congestion increases with the increase in the number of nodes sending the event information (Tilak, Abu-Ghazaleh, & Heinzelman, 2002), resulting into high degree of packet and energy loss.

In WSANs, the information is transported to multiple destinations (actors) as compared to a single destination (sink) in sensor networks. Hence, apart from above mentioned issues of transport layer, selection of an appropriate actor prior to information transport is necessary. A high degree of actor to actor coordination is required, for the selection of a suitable actor, which can be one of the many actors that are deployed (at different locations) in the sensor field (Akyildiz & Kasimoglu, 2004).

2.3 Traditional Transport Protocols and Sensor Networks

2.3.1 Transmission Control Protocol

Transmission Control Protocol (TCP) (Postel, 1981) is the most well-known transport protocol. TCP uses a connection-oriented approach with end-to-end acknowledgements (ACKs) and retransmission to guarantee reliability.

As described in section 2.2, wireless sensor networks do not require guaranteed reliability for sensors-to-destination transport due to the presence of redundant information, energy conservation and application dependent nature of these networks (Vuran, Akan, & Akyildiz, 2004). TCP is connection oriented transport protocol in which data transport starts after a three-way handshake process.

In wireless sensor networks, sensor nodes transmit event information (some value of interest) to a sink that is not more than several bytes (Wang, Sohraby, Hu, Li, & Tang, 2005). Thus, implementing a handshake process for such small size data is a big overhead and consumes considerable energy. Wireless links are prone to failure due to environmental conditions and low power transmission mode used by sensor nodes for energy conservation (Zhao & Govindan, 2003). Hence, the connection setup process can be more time consuming than in wired networks.

TCP can be considered for destinations-to-sensors transport in wireless sensors networks but the following observations show that TCP is also not suitable for destinations-to-sensors transport.

- TCP shows degraded performance in heterogeneous networks that comprise of wireless links. This is because, TCP considers packet loss as a sign of congestion not the lossy wireless links, but in fact lossy links are the major source of packet loss in wireless networks.

Findings of Balakrishna, Padmanabhan, Seshan, & Katz (1997) and Chaskar, Lakshman, & Madhow (1999), demonstrate the poor performance of TCP on wireless links.

- Wan, Campbell, & Krishnamurthy (2005), shows that an end-to-end transport solution is not feasible for wireless sensor networks due to high channel error rate and multi-hop transmission in these networks. According to the findings of these authors, end-to-end reliable packet delivery ratio decrease below 50% under a uniform channel error rate of 20% in only a four hop wireless sensor networks.
- TCP uses end-to-end ACK and retransmission to guarantee reliability. This approach cause much lower throughput and longer transmission time if RTT (Round-Trip Time) is larger as that in large-scale WSNs, since the sender will stop to wait for the ACK after each data transmission (Wang et al., 2005).
- Due to small memory size and limited energy resource of sensor nodes TCP is not a good candidate for sensor networks due to it is computational complexity (Chong & Kumar, 2003).

2.3.2 User Datagram Protocol

User datagram protocol (UDP) (Postel, 1980) is a connectionless transport control protocol. According to the findings of Wang et al. (2005), UDP is not suitable for WSNs due to the following reasons:

- There is no flow control and congestion control mechanism in UDP. If UDP is used for WSNs, it will cause lots of datagram dropping when congestion happens. In this point at least, UDP is not energy-efficient for WSNs.
- UDP contains no ACK mechanism, no any reliability mechanism. The datagram loss can be only recovered by lower MAC algorithms or upper layers including application layer.

2.3.3 Reliable Multicast Protocols

Multicast transport protocols based on UDP have been studied in fare detail. Multicast protocols like, reliable multicast transport protocol (RMTP) (Lin & Paul, 1997) and scalable reliable multicast (SRM) (Floyd, Jacobson, Liu, Macanne, & Zhang, 1997) provide good concept for a transport mechanism that could be used for sensor networks; especially for destination(s)-to-sensor transport. For example, SRM provides a guaranteed delivery of sequenced data to a multicast group and avoids ACK implosion using NACKs. NACKs are multicast so that any receiver which has the missing fragments cached can provide those. However, SRM represents a traditional receiver-based reliable transport solution and is designed to be highly scalable for internet applications. But, SRM is designed to operate in a transport medium is highly reliable (wired internet) and does not suffer from the unique problems found in wireless sensor networks, such as, hidden terminal and interference.

Like other transport protocols for wired and wireless networks, the major problem with multicast transport protocols is that they are not designed keeping in mind the energy constraints of sensor networks (Karl & Willig, 2005).

2.4 Information Flows in Sensor Networks

The research in the field of reliable transport in sensor networks can be categorized by the flow of information in these networks.

2.4.1 Sensors-to-Destination(s) Flow

Sensor nodes while performing sensing task can send information to a single or multiple destinations periodically, on request and on event occurrence. In order to conserve network energy, it is required in most applications, that nodes only send information when an event occurs (Akan & Akyildiz, 2005). The flow of information in case of WSNs is to a single destination (sink) while in case of WSANs can be to multiple destinations (actors). The nodes-to-destination flow in sensor networks is also termed as *many-to-one*, *upstream*, *sensors-to-sink*, *sensor-to-destination* and *event flow* (Akan & Akyildiz, 2005; Gungor & Akan, 2007; Yangfan, Micheal, Jiangchuan, & Hui, 2005).

2.4.2 Destination-to-Sensors Flow

The large scale and random deployment of sensor networks demands for reliable information transport from destination (sink/actor) to sensor nodes. The basic reason for this information transport include updating of event definitions on sensor nodes, complete change of binary codes of sensor nodes and occasionally for network status monitoring (Wan, Campbell, & Krishnamurthy, 2005). The flow of information from a sink to the nodes is termed as *destination-to-nodes*, *one-to-many*, *downstream* and *sink-to-nodes* flow in sensor networks. In case of WSANs, destination-to-sensors flow is triggered by the sink, instructing the actors to send necessary information to sensors.

2.5 Sensors-to-Destination Reliable Transport and Event Reporting

Considerable amount of research in the field of reliable sensors-to-destination transport and event reporting has been done, in the last decade (Ilyas & Mahqoub, 2004; Wang & et al. 2005). The focus of these protocols is to ensure an increase in successful delivery of event packets or other information transmitted by sensors to destination (sink/actor). In order to ensure that event packets must reach the destination and are not dropped, these protocols generally implement congestion detection, avoidance or mitigation schemes along with rate control mechanisms. The primary design constraint of these protocols is to conserve energy either by avoiding or removing congestion. Also energy is conserved, by not increasing the reporting rate of nodes, once required throughput for successful event detection is achieved at the destination.

In contrast to the normal event reporting or transport, some applications (Cerpa & Estrin, 2002; Cook & Das, 2004) require a periodic/continuous feed back from the nodes; in order to have an up to date picture of the sensed environment. In sensor networks, nodes that are near to the sink can communicate more easily with the destination than the nodes that are farther away from destination. When most of the nodes are sending data to destination, network becomes congested and nodes that are far away are more affected than nodes nearer to the destination. This requires regulating all nodes in the network in such a way that every transmitting node should get a portion of the network bandwidth, resulting into *fairness* (Tien & Bajcsy, 2004). Also, some protocols have been proposed for important quality of service (QoS) issues that some application requires in wireless sensor networks. These issues include reliable event reporting for multiple events and time-bound event reporting (Gungor & Akan, 2007).

A detailed survey of existing event reporting and reliable transport protocols in WSNs and WSANs are given below. The terminology *upstream reliability* or *upstream transport* is used to refer to the direction of information flow.

- Reliable Multi-Segment Transport (RMST) (Stann & Heidemann, 2003) provides a transport mechanism for wireless sensor networks. RMST is specifically designed to work over the directed diffusion (Intanagonwiwat & et al., 2002); routing layer. RMST is designed for delivering larger blocks of data in multiple segments from a source node to a sink node. It is a selective NACK-based protocol that can be configured for in-network caching and repair. In RMST a unique *entity* is a data set consisting of one or more fragments coming from the same source. Reliability in RMST refers to the eventual delivery of any/all fragments coming from a unique entity to all subscribing sinks.

  RMST provides mechanisms both for in-network caching (hop-by-hop) and with out in-network caching (end-to-end). However, best results are achieved with in-network caching in which each intermediate hop caches the fragments to observe *holes* (missing fragments). If holes exist they are reported to the upper node, this continues up to the sender until it is retransmitted. In the end-to-end scenario the destination will send the NACK on the reverse path to the sender (taking advantage of directed diffusion's fixed paths). RMST suggests that reliability both at Medium Access Control (MAC) and transport layer is important. MAC level reliability is important not only to provide hop-by-hop error recovery for the transport layer, but also for route discovery and maintenance. Hence, RMST provides best results when used with a selective ARQ (Automatic Repeat reQuest).

- Event to Sink Reliable Transport (ESRT) (Akan & Akyildiz, 2005) provides an event transport mechanism, which is controlled by the sink. It is based on the fact that sensor networks are generally deployed to observe events and critical events must be reliably transported. ESRT measures reliability in terms of number of packets received at the sink during an interval (maintained at sink) termed as *observed event reliability*. The required level of reliability is application defined and is termed as *desired event reliability*. ESRT also calculates an *optimal reporting frequency* for the network after which increasing reporting rate of nodes results into congestion. So, optimal frequency is used for congestion avoidance. In order to achieve desired reliability, after the end of each interval the sink decides to increase or decrease the reporting rate of the nodes based on the observed event reliability level and current reporting frequency of nodes.

In ESRT nodes monitors their local buffer level to predict for congestion in the next interval. If a node observes that during next interval it will be congested it informs the sink (by setting a bit in the forwarding packet) which decreases the reporting rate of the network. ESRT assumes that the sink broadcasts the reporting frequency at high energy so that all the nodes can hear it; which might not be possible for large-scale networks and it can also interfere with normal transmissions (Yangfan & et al., 2005). Also, the congestion control mechanism of ESRT always regulates all the sources; regardless of where the congestion occurs.

- Price-Oriented Reliable Transport (PORT) (Yangfan & et al., 2005) protocol for wireless sensor networks facilitates sink to achieve reliability. The authors suggest that packets from different sources may have different contribution to improve sink's information on the phenomenon of interest. Communication costs between sources and the sink may be different and may change dynamically. Therefore, authors discuss that reliability can not be simply measured by the total incoming packet rate at the sink. PORT defines sensor to sink data transport to be reliable when the transport mechanism can assure that the sink can obtain enough fidelity of the knowledge on the phenomenon of interest.

  Each node in PORT calculates a price, which is equal to total number of transmission attempts made by all in-network nodes for successful delivery of a packet, from a source node. Since, the price increases with increase in congestion on a route, nodes with lower price are preferred to report events with high reporting rates. In PORT, sink directs individual nodes to increase or decrease their reporting rates by sending control information. In dense networks, since nodes can be at multiple hop distance from the sink, sending such control information to every node separately is very difficult (Hussain, Seckin, & Cebi, 2007).
- Interference-aware fair rate control in wireless sensor networks, (IFRC) (Rangwala, Gummadi, Govindan, & Psounis, 2006) monitors average queue size to detect incipient congestion and uses Additive Increase Multiplicative Decrease (AIMD) scheme to adjust the reporting rate of nodes. IFRC does not imply strict fairness and allows flows passing through less restrictive contention domains to have higher rates than the ones passing through higher contention domains. IFRC considers a tree-based architecture of nodes in which nodes avoid packet drops by identifying potential interferers for each node. A potential interferer of a node includes not only the neighboring (first hop) nodes but also the neighboring node's neighbors too. Nodes share their congestion information with all of their potential interferers. According to the congestion status of potential interferes, nodes dynamically adjust their

reporting rates. Since IFRC only takes effect after congestion happen, it cannot mitigate congestion and avoid packet drops (Shanshan, Xiangke, Shaoliang, Peidong, & Jie, 2007).

- Credit based fairness control in wireless sensor networks (CFRC) (Shanshan & et al., 2007) proposes a mechanism to ensure that all data sources have equal or weighted access to end-to-end network bandwidth. CFRC allocates bandwidth to nodes based on *credit*; the effective amount of sensed information, which is dependent on node density and their distribution instead of uniformity. In CFRC, all nodes including congested nodes allocate bandwidth to their upstream neighbors according to the *credit* of each upstream neighbor. Aggregation nodes (intermediate nodes) in CFRC, computes the credit of aggregated packets using simple sum operation, the collective outcome ensures that data sources share weighted downstream bottleneck bandwidth.
- Congestion Control and Fairness for many-to-one routing in sensor networks (CCF) (Tien & Bajcsy, 2004) proposes an algorithm that ensure fairness by assuming that all the nodes are transmitting and routing data at the same time. CCF uses buffer size to detect for congestion. CCF implements a tree based technique in which each node calculates its sub-tree size. Reporting rate is allocated to nodes depending on their sub-tree sizes. Every node maintains a separate queue for each of their previous hop nodes. In order to ensure fairness, nodes forward packets from these queues depending on the sub-tree size of the previous hop nodes during each epoch.

According to Shanshan & et al., (2007), in CCF each sensor allocates bandwidth only based on the size of its sub-tree and hasn't considered the effect of other interferers to congested node. Rangwala et al., (2006), suggests that CCF provides low throughput since it selects a fix length epoch for forwarding packets which is not dependent on network conditions. According to Hussain, Seckin, & Cebi, (2007), sensor nodes have limited memory resources, maintaining a separate fixed size queues for each previous hop node is not a memory efficient solution; especially in dense networks. In case of multiple events, CCF treats all events similarly which can have different reporting rate requirements.

- Delay-aware reliable transport (DART) in wireless sensor networks (Gungor, & Akan, 2007) aims to provide time-bound and reliable event transport from the sensor field to the sink with minimum energy consumption. DART defines transport to be reliable and delay-aware if the packets are received within application defined time bound and at application defined reporting rate. DART uses time critical event packet scheduling policy to forward packets according to their deadlines. Sink-based rate control and congestion mitigation scheme is used in DART, in which the sink adjusts the reporting rate of the event region after periodic intervals.
- Melodia, Pompili, Gungor, & Akyildiz, (2005), propose a distributed coordination framework for wireless sensor and actor networks. A new sensor-actor coordination model is proposed, based on an *event-driven clustering* paradigm in which cluster formation is triggered by an event. Hence, clusters are created on-the-fly for optimally reacting to the event itself and providing the required reliability with minimum energy expenditure. A model for actor-actor coordination is introduced for a class of coordination problems, according to that, the area to be acted upon is optimally split among different actors.
- Shah, Bozyigit, Hussain, & Akan, (2006), present a multi-event adaptive real-time coordination and routing mechanism for in wireless sensor and actor networks. The framework forms clusters which are adaptive to the nodes energy and their multiple events reporting rate. It addresses the issues of nodes heterogeneity, real-time event delivery and coordination among sensor-sensor, sensor-actor and actor-actor. Only the cluster-heads coordinate with the interested client (sink/actors) in order to achieve energy efficiency.

2.6 Destination-to-Sensors Reliable Transport in Sensor Networks

A number of transport protocols (Wan, Campbell, & Krishnamurthy, 2005; Park, Vedantham, Sivakumar, Akyildiz, 2008) have been proposed for Sink-to-Node(s) transport in wireless sensor networks. All these protocols use some form of in-network caching and a hop-by-hop transport mechanism. According to Wan, Campbell, & Krishnamurthy, (2005), the reason for this approach is that, the hop-by-hop scheme divides the typical multi-hop forwarding operation into series of single hop transmissions. In case of a packet/fragment loss to an intermediate node the probability of loss detection is higher and the packet loss will be immediately detected by the intermediate node. On the other hand, an end-to-end transport scheme can only detect the packet loss at the final destination. Another reason that supports the use of hop-by-hop mechansims is that, sink-to-node(s) transport is used for application such as re-tasking/reprogramming which involves the whole network or a group of nodes. Therefore, the cost of transmitting data through the intermediate nodes is either zero (whole network) or minimal (Wang, & et al., 2005). Some commonly proposed sink-to-nodes transport protocols are given below:

- Pump Slowly Fetch Quickly (PSFQ) (Wan, Campbell, & Krishnamurthy, 2005) is a sink-to-nodes transport protocol for wireless sensor networks. PSFQ uses controlled flooding and stop-and-forward transport mechanism. PSFQ comprises of three functions: message relaying (pump operation), relay-initiated error recovery (fetch operation) and selective status reporting (report operation).

  Pump operation is basically restricted flooding in which a node broadcasts packets to its neighbors at a slow rate (compare to fetch operation). The pump operation operates in a multi-hop packet forwarding mode and the nodes use stop and forward mechanism to ensure ordered delivery of fragments. *Fetch operation* can be triggered by a node, once a sequence number gap in the fragments is found. In fetch mode, a node aggressively broadcasts NACK messages to its neighbors (containing missing sequence numbers). If no reply is heard until the fetch timer expiry (much smaller than pump timer), it repeatedly sends NACKs for some times. *Report operation* can only be activated by the sink node which sends a report message to nodes in the network using a destination ID or hop number. The particular nodes broadcasts back the transport-report to its neighbors. Each neighbor appends its report to the existing report. According to Park, & et al., 2008, PSFQ is a NACK based protocol which can not ensure single-packet delivery. Moreover, PSFQ uses a fixed channel error model and requires fine tuning of timers according to network conditions. Also, PSFQ increases the latency of delivery in-order to decrease the energy consumption.

- GARUDA (Park, & et al., 2008) is an approach for reliable downstream data delivery in wireless sensor networks. For every new message (e.g., file) to be transmitted by the sink, GARUDA requires small finite series of short duration pulses (twice the amplitude of normal transmissions) to be propagated periodically through out the network. These pulses are used to ensure first packet delivery and for the creation of core (a backbone for communication). The core is constructed during this first packet flood assuming a simple 100 percent network wide reliable flood. The core is comprised of nodes that are at $3n$ (where $n = 1, 2, 3 \ldots$) hop distance from the sink. Every $3n$ hop node selects its self to be a core node, if it does not hear from any other node in its band ($3n$).

In order to increase the channel utilization, GARUDA supports *out of sequence* packet delivery among the core nodes, requiring them to exchange A-Map (availability map) information; on the cost of increased energy consumption. The intermediate nodes hear the transmission of core nodes to get missing packets. Exchange of A-map to update neighboring nodes about the status of packet delivery imposes a considerable overhead. Apart from that WFP pulses can interfere with normal transmissions. According to Vedantham, Sivakumar, & Park, (2007), buffer overflows are more likely to happen in *out of sequence* packet delivery case, however GARUDA do not address this issue. GARUDA can consume more energy due to WFP pulses, A-map exchange and overhearing of intermediate nodes especially in dense networks.

- A reliable transport protocol (ATP) is a new transport protocol for ad hoc networks (Sundaresan, Anantharaman, Hseeh, & Sivakumar, 2005). It is a receiver-based and network-assisted end-to-end feedback control algorithm. It uses selective ACKs (SACKs) for packets loss recovery. In ATP, intermediate network nodes compute the sum of exponentially averaged packet queuing delay and transmission delay, called *D*. The idea is that the required end-to-end rate should be the reverse of *D*. The *D* is computed over all the packets traversing the node and used to update the value piggybacked in each outgoing packet if the new value of *D* is bigger than the old value. After this hop-by-hop computation and piggyback, the receiver can get the largest value of *D* that each packet experience on the way. Then the receiver can calculate the required end-to-end rate, the reverse of *D*, for the sender and feedback it to the sender. Then the sender can intelligently adjust its sending rate according to received *D* from the receiver. In order to guarantee reliability.

ATP uses selective ACKs (SACKs) as an end-to-end mechanism for loss detection. But the SACK block in ATP is 20, much larger that that in TCP (only 3). ATP decouples congestion control and reliability and achieves better fairness and higher throughput than TCP. ATP doesn't consider energy issues and its end-to-end approach might be not the optimal for sensor networks (Wang & et al., 2005).

2.7 Congestion Avoidance and Control in Sensor Networks

Wireless sensor networks generally use radio transmission for data dissemination. Therefore, the basic sources of congestion in these networks are the nature of lossy radio links, collisions/interference and congestion due the activation of a group of nodes in case of an event (Akan & Akyildiz, 2005; Karl & Willig, 2005). On event occurrence suddenly data starts flowing from the event nodes which results into congestion; buffer overflows. Routing layer and the MAC layer can take joint actions to avoid routing data to these lossy links (Al-Karaki & Kamal, 2004). The MAC layer (both contention and TDMA based) is responsible for medium access; therefore reducing collisions is the duty of MAC layer (Chlamtac, Conti, & Liu, 2003). According to Akan & Akyildiz, (2005), the transport layer should handle congestion occurring due to an event occurrence or congestion due to large data transfers e.g., image or binary code. Commonly used congestion detection and mitigation protocols for sensor networks are given below:

- Congestion detection and avoidance (CODA) (Wan, & et al., 2003) protocol is based on event-driven sensor networks which operate under idle or light load. But when an event occurs, sensors suddenly become active and large event impulses generally result in congesting the network. CODA uses channel sampling and buffer occupancy as the basic metrics for the detection of congestion. Channel is only sampled at periodic intervals when the buffer occupancy is above a certain threshold value; for decreasing the energy consumption. CODA employs open-loop hop-by-hop backpressure and closed-loop multi-source regulation schemes for mitigating congestion. *Open-loop, hop-by-hop backpressure* deals with transient holes (temporary congestion areas) which can occur near the source or further away from it. Once congestion is detected by a node backpressure messages are broadcasted to the neighbor nodes. These messages travel upstream towards the source. An intermediate node depending on its buffer occupancy and traffic monitoring statistics decides to further propagate these messages or to stop propagating them.

  Closed loop, multi-source regulations deals with persistent congestion in the network. The source only enters in sink regulation mode if the source event rate exceeds the theoretical maximum throughput of the channel. As a result, the source is more likely to contribute to congestion and therefore closed-loop control is triggered. At this point a source requires constant feedback (e.g., ACKs) from the sink to maintain its reporting rate. According to Hu, Xue, Li, Xie, & Yang, (2005), the open loop hop-by-hop mechanism of CODA, decreases the sending rate of the upstream nodes according to the depth of congestion which is not increased after congestion is mitigated.

- SenTCP (Hu, & et al., 2005) is a congestion control protocol for wireless sensor networks. It uses hop-by-hop, open loop congestion control mechanism. It detects and avoids congestion using both buffer occupancy and packet inter-arrival time. SenTCP focuses only on congestion control not on loss recovery. Like CODA, it considers event impulses as the basic reason for congestion. CODA issues feed back signals when buffer occupancy and/or channel load overruns a threshold; so they are used for reducing sending rate (in the open loop mechanism). On the other hand, SenTCP uses periodic feed back signals to adjust (increase/decrease) the reporting rate of upstream nodes; according to their local congestion status. SenTCP avoids congestion by maintaining the reporting rate of nodes below channel threshold and reducing sending rate if the neighboring sensor nodes have large occupied buffer ratio.
- Priority-based congestion control in wireless sensor networks (PCCP) (Wang, Li, Sohraby, Daneshmand, & Hu, 2007) uses packet inter-arrival time and packet service time to detect congestion level at a node and employs weighted fairness to allow nodes to receive priority-dependent throughput. PCCP suggests that sensor nodes might have different priority due to their function or location. Therefore, nodes with higher priority-index gets more share of the bandwidth in order to ensure priority dependent throughput. The priority-based rate adjustment scheme of PCCP uses congestion degree and priority index of a node to adjust its reporting rate. CODA (Wan, & et al., 2003), SenTCP (Hu, & et al., 2005) and PCCP (Wang, & et al., 2007) use source based congestion in which congestion signals propagate back from the congestion region to the source nodes.

According to the findings of Hussain, Seckin, & Cebi (2007), source-based congestion mitigation techniques in dense networks is not a good solution. Because of high node density, these congestion signals are dropped and they do not reach to source nodes.

- Mitigating congestion in wireless sensor networks (Hull, Jamieson, & Balakrishnan, 2004), proposes three techniques that span on different layers of the traditional protocol stack: hop-by-hop flow control (based on buffer occupancy), rate limiting to implement fairness and a prioritized medium access control (MAC) protocol.

First, hop-by-hop flow control that resembles the backpressure mechanism of CODA (Wan, & et al., 2003) but it replaces the explicit control packets with a piggybacked congestion bit carried by all packets. In order to detect congestion at a neighboring node, a node overhears all neighboring nodes transmissions. If a packet with congestion bit set is received from a neighboring node, the node will stop its transmission until congestion mitigates at the neighboring node.

Second rate limiting, a node is required to listen to its parent's transmission to estimate for the total number of unique sources (N) routing through the parent. It then uses a token bucket scheme to regulate each sensor's send rate. A node is allowed to send if its token count is above zero and each send costs one token. The token bucket scheme rate-limits the sensor nodes, in order to send packets according to the rates of each of its descendent. This scheme is applicable for a network in which nodes offer same traffic load and the routing tree is not significantly skewed.

Third prioritized MAC solution, it decreases the back-off window of a congested node to one fourth the size of a non-congested node, allowing the congested node to get more access to the medium.

- Price-oriented reliable transport protocol for wireless sensor networks, (PORT) (Yangfan et al., 2005) uses link loss estimation as a basic source of congestion detection and avoids congestion by dynamically forwarding packets to less congested nodes. In dense networks, link losses are high which are generally not because of congestion but due to packet collision (Hussain, Seckin, & Cebi, 2007). In PORT, sink directs individual nodes to increase or decrease their reporting rates. However in dense networks, sending such control information to every node is very difficult, since nodes can be at multiple hop distance from the sink.

- Interference-aware fair rate control in wireless sensor networks, (IFRC) (Rangwala et al., 2006) detects congestion by monitoring average queue length and exchanges congestion state among the potential interferers using a congestion sharing mechanism. In IFRC each node adds its buffer size and current congestion state in every packet that it forwards resulting into extra energy consumption on per packet basis (Wang, & et al., 2007).
- Shigang & Na, (2006) presents congestion avoidance based on light-weight buffer management in wireless sensor networks. Their work is impressed by the idea of flow control in ATM (Asynchronous Transfer Mode) networks proposed by Kung, Blackwell, & Chapman, (1994), which suggests that a sender should transmit a packet only when it knows that the receiver has the buffer to store the packet. Light-weight buffer management is proposed for both CSMA (Carrier Sense Multiple Access) and TDMA (Time Division Multiple Access) based medium access protocols (MAC). For both MAC protocols, data packets are piggybacked to update buffer state. When a sensor *x* sends out a data packet, it piggybacks its residual-buffer size in the frame header. If a neighbor *y* overhears a frame from *x*, it caches the residual-buffer size of *x*. When *y* overhears a packet that is sent by another sensor to *x*, it reduces the residual-buffer size of x by one.
- Galluccio, Campbell, & Palazzo (2005), propose an aggregation-based congestion control for sensor networks (CONCERT). The authors of CONCERT suggest the use of adaptive data aggregation in order to reduce the amount of information traveling through out the network rather than using a back-pressure approach to regulate source nodes transmission rate on congestion. CONCERT uses data aggregator nodes to do the data aggregation, which are congestion prone nodes in the network. Aggregator

nodes depending on the degree of congestion aggregates the incoming data packets in order to avoid buffer overflow.

- Congestion control from sink to sensors (CONSISE) (Vedantham, Sivakumar, & Park, 2007) adjusts the downstream sending rate at each of the sensor nodes to utilize the available bandwidth depending on the congestion level in the local environment. The authors suggest that downstream information flow can also result into congestion, similar to upstream information flow. CONSISE describes basic reasons of downstream congestion as reverse path traffic and broadcast storm problem. Therefore, a node in CONSISE protocol using incoming traffic rate and out going traffic rate during a small time interval (epoch), predicts for congestion level and adjusts the reporting rate.

CHAPTER THREE

WIRELESS SENSOR AND ACTOR NETWORKS

3.1 Architecture of Wireless Sensor and Actor Networks

Wireless sensor and actor networks (WSANs) are application dependent networks therefore the arrangement of nodes, actors and sink is also application dependent. The operational architecture of these networks can be categorized as automated or semi-automated (Akyildiz, & Kasimoglu, 2004), according to the information flow from sensor nodes to either actors or sink.

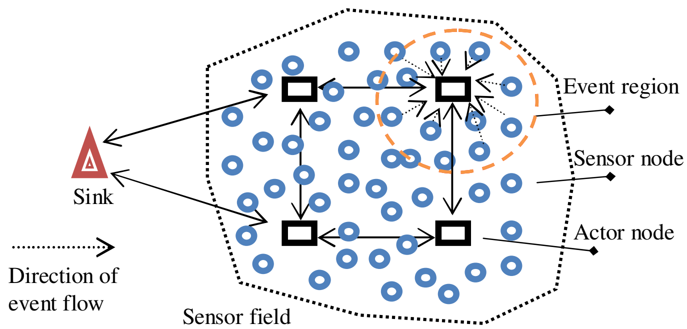

Figure 3.1 Automated wireless sensor and actor network architecture.

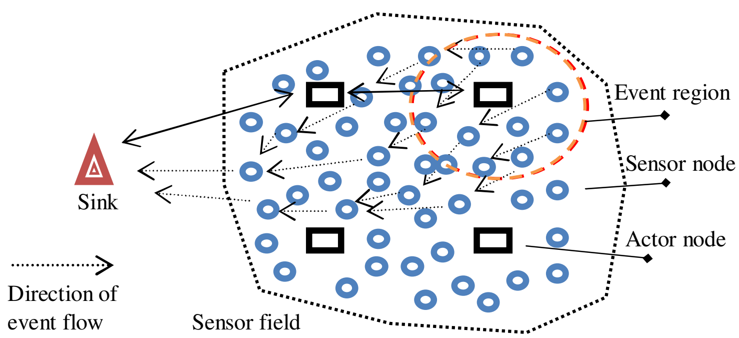

Figure 3.2 Semi-Automated wireless sensor and actor network architecture.

In automated WSANs (Figure 3.1), nodes send event or sensing information to the actor nodes which take appropriate action. In this architecture, the sink is generally not involved in decision making process. Sink controls the overall communication and the external entity (user) interacts with the sink for controlling or querying the network. In semi-automated WSANs (Figure 3.2), nodes send the event or sensed information to the sink node which performs the decision making process (generally external entity is involved) and it activates the actor(s).

In both architectures, depending on application scenario and node capabilities, a sink can directly communicate to actors or it may use intermediate nodes to rely the information. Also the actors may be able to directly communicate to each other or intermediate sensor nodes may be used for relying. The advantage of semi-automated architecture is that, it is similar to the one being used for WSNs, but to achieve quick response-time and longer network lifetime automated architecture is more suitable. Therefore, the term Wireless Sensor and Actor Networks (WSANs), generally refers to automated WSAN's architecture, as shown in Figure 3.1.

3.2 Application Areas for Wireless Sensor and Actor Networks

Sensor networks consists of many different types of sensors such as seismic, thermal, visual, infrared, acoustic low sampling rate magnetic and radar, which are able to monitor a wide variety of ambient conditions. Some of these conditions are listed below:

- Temperature
- Humidity
- Vehicular movement
- Lightning condition
- Pressure
- Soil makeup
- Noise levels
- The presence or absence of certain kinds of objects
- Mechanical stress levels on attached objects
- Characteristics such as speed, direction, and size of an object

The rapid deployment, self-organization and fault tolerance characteristics of sensor and actor networks make them very promising systems for different application domains. Some of the important application areas of sensor networks are given below:

- Military applications: movement of friendly forces, battle damage assessment, target tracking, nuclear, biological and chemical attack detection, disaster recovery (Cook, & Das, 2004).
- Environmental applications: tracking the movements of birds, small animals, and insects, monitoring environmental conditions that affect crops and livestock, irrigation, macro-instruments, flood detection and forest fire detection (Cerpa, & et al., 2000; Essa, 2000).
- Health applications: integrated patient monitoring, diagnostics and drug administration in hospitals, monitoring of human physiological data and tracking and monitoring doctors and patients inside a hospital (Coyle, Boydel, & Brown, 1995; Johnson, & Andrews, 1996).
- Home applications: light, temperature and microclimate control, intelligent home devices like vacuums and fridges, intrusion detection (Essa, 2000).
- Commercial applications: monitoring material fatigue; managing inventory; monitoring product quality; detecting and monitoring car thefts; vehicle tracking and detection (Cook, & Das, 2004).

3.2.1 Wireless Sensor and Actor Networks for Mining

This study presents an application of WSANs in the mining field. The architecture of the application is used to define the basic consideration of this study. Also, the need for a reliable data transport in WSANs is highlighted using this application. However, the reliable transport mechanism presented in this study is not limited to mining application only. It is equally viable for different applications especially general disaster recovery, environment monitoring and control applications.

In the mining application, a wireless sensor and actor network is deployed in a mine for the purpose of monitoring environmental conditions and to prevent and recover from mine disasters. The network can perform following functions:

1. Monitoring environmental conditions inside a mine. For example, temperature, humidity, pressure and oxygen content in air etc.
2. Providing quick relief by triggering actors (alarms), during disasters like fire, or leakage of poisonous gases.
3. Finding location of trapped miners within the mine in case of mine collapses.

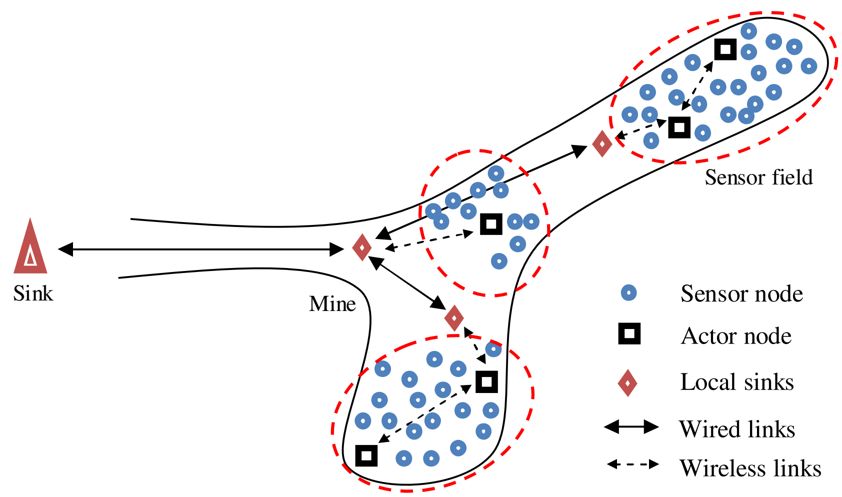

Figure 3.3 Wireless sensor and actor network's architecture for mining application.

The architecture of the application shown in Figure 3.3, it consists of sensor and actor nodes placed underground inside the mine while the sink remains outside the mine. The actors (e.g., alarms) are energy rich devices which communicate with the local sinks using wireless communication. The local sinks are simple information routers which are connected to other local sinks and the main sink outside of the mine, using wired or wireless communication media. The local sinks act as dummy sinks for the actors, so that they communicate with their nearby local sink just like the main sink. The sensor nodes are thrown or scattered in the mine. Areas of interest in mines are the regions where miners are working. Therefore, deploying nodes throughout the mine with same density is not a cost-effective solution. Instead, density is high in regions where miners are working or regions that are more critical.

Mining application uses an automated architecture for reporting events. The information from the sensors is transmitted to nearby actor which can take localized action. The actors transmit event information received from sensors, to the sink to update sink's status of the network. Actor-to-actor communication in WSANs is similar to the communication paradigm of ad hoc networks, due to the small number of resource-rich actor nodes being loosely deployed.

In the related literature, there are several transport protocols dealing with ad hoc networks (Chlamtac, Conti, & Liu, 2003). In this study, no transport solution has been proposed for actor to actor coordination. Likewise, actor to local sink communication is similar to ad hoc networks communication paradigm and is not considered in this study. The local sinks are connected to the main sink out side the mine with either wired or wireless communication media.

3.3 Need for a Reliable Transport Solution in Wireless Sensor and Actor Networks

The need for a reliable transport solution has been identified in a number of existing works (Akyildiz & et al., 2002; Pottie & Kaiser, 2000; Tilak, Abu-Ghazaleh, & Heinzelman, 2002). The importance and need for an information transport mechanism in WSANs, with respect to the mining application is given below:

- The sudden increase in the temperature of a certain region within a mine (possibly fire) must be quickly and reliably informed to the actors to trigger the fire alarms. So, that the miners can be timely evacuated.
- The WSAN in mining application observes different events. Some of these events are inter-related e.g., fire and oxygen content in air. Hence, multiple events can be triggered at the same time. Reliable transport of multiple events

is required in the mining application, in which one event can demand higher reporting rate while the other event can have considerably low reporting rate.

- In sensor networks the flow of information from nodes to destination suffers congestion as result nodes nearer to the destination sends more event information to destination than the nodes farther away from destination. This decreases the destination's visibility in terms of overall status of network. For events like leakage of poisonous gas, the actors/sink require precise per node information in order to identify unsafe regions in the mine. Hence, information transport with fairness is also critical.
- Sensor nodes not only report events but also send periodic sensed data to the destination as programmed. The sensed data in terms of position of an object can be time-dependent and can have certain time bounds. So the transport of time-critical information reliably with in certain bounds is another issue of transport in WSANS.
- A reliable transport mechanism is required for the re-tasking of nodes, in case of a disaster e.g., mine collapse, for the purpose of finding entrapped miners.
- Since the network can operate for a long period possibly for months to year(s), it might be required to reprogram the nodes completely. Reprogramming requires transport of new binary files from the sink to sensors.
- Nodes on the occurrence of an event compares the event values e.g., temperature, with an event-database in order to determine the criticality of the event. If the event is critical, then it is reported to the destination otherwise it is ignored. This small event-database can contain from few integers to image files (according to event needs). The user might require updating the event-database, reliable transport is required for this purpose, so that the whole network or a particular sub-region can be updated.

3.4 Design Challenges for Transport Layer in Wireless Sensor and Actor Networks

WSANs are characterized by their unique challenges which distinguish these networks from other wired and wireless networks. These constraints affect the design of communication protocols for these networks. Following are some of the basic characteristics of WSANs and their impact on the transport layer design.

- *Random deployment:* Nodes in WSNs are generally thrown or scattered in the sensing field without any fixed topology. Although, physical placement primarily determines connectivity but different problems, such as obstructions, interference, environmental factors, antenna orientation, and mobility make connectivity difficult to maintain (Ilyas, & Mahqoub, 2004; Rappapport, 2002). Due to these problems, the network is required to periodically discover and adapt to presently available connectivity.
- *Scalability and density:* Hundreds to several thousands of nodes can be deployed throughout the sensor field, depending on the application's nature. Due to limited radio range and the possibility of node failures, they are deployed within tens of feet of each other (Intanagonwiwat, & et al., 2002). The node densities may be as high as 20 nodes/m3 (Shih, & et al., 2001). Another work by Cho, & Chandrakasan, (2001), state that density can range from few sensor nodes to few hundred sensor nodes in a region, which can be less than 10m in diameter. Densely deployed high numbers of nodes, require careful handling and maintenance of network topology. During event transport, the degree of congestion increases with the increase in node density (Gungor, & Akan, 2007; Hussain, Seckin, & Cebi, 2007).
- *Unattended operation:* Sensor nodes operate without any human interference. Due to large scale deployment and high density, it is generally not possible to physically locate a node in case of failure. This implies that sensor network protocols and algorithms must possess self-organizing capabilities in order to establish and maintain network. With respect to transport protocols, this

feature indicates the importance of reliable sink-to-sensors transport, as data from a single source needs to be transported to hundred and thousands of nodes.

- *Radio links and high channel error rates:* Radios of sensor nodes generally operate at a frequency range of 916/433 MHz; the ISM (industrial, scientific and medical) radio bands. Radios consume approximately 25mA/10mA current for transmission/reception respectively (Gardner, Varadan, & Awadelkarim, 2001). The radio range is generally measured in tens of meters. Due to low lying wireless antennas, communication is affected by scattering, shadowing, reflection, diffraction, multi-path and fading effects (Hashemi, 1993; Rappapport, 2002). As a result the channel error rate is generally high and variable in nature.

  Channel error rates in closed environments such as mining application will be especially high due to multi-path effects which are evident due to reflections from the floor and walls of the mines. According to the findings of Zhao, & Govindan, (2003), in WSANs, the wireless link quality between pairs of nodes varies during the lifetime of a network based on distance, transmit power, radio interference, and environmental factors such as obstructions (walls or rocks) and people in the sensor network field attenuating radio signals.

- *Limited Energy resource:* Sensor nodes are generally powered by standard batteries (<0.5 Ah, 1.2 V) (Gardner, Varadan, & Awadelkarim, 2001). The most important challenge in sensor networks is energy saving. Providing a transport layer solution with low latency, better channel utilization and with minimum energy consumption is an uphill task.

CHAPTER FOUR

PROPOSED TRANSPORT LAYER FOR WIRELESS SENSOR AND ACTOR NETWORKS

4.1 Introduction

A scalable, energy-aware and flexible transport layer for wireless sensor and actor networks (WSANs) is presented in this study. The solution encompasses reliable information transport from actors-to-sensors, sensors-to-actors with a congestion control scheme.

Scalability in this study is defined in terms of number of nodes within the network or number of nodes attached (reporting event) to a single actor. This can range from a few nodes to tens of nodes reporting information to a single actor.

Energy-aware refers to the fact that the transport solution minimizes packet losses due to congestion, interference and does not increase the reporting rate of nodes more than required by application. Thus, saving energy of nodes and extending the life time of network. The proposed transport layer not only provides simple information transport form sensor nodes to actors but also handles transport of multiple events and time-critical events. Also, for periodic information gathering from sensors or for events, that require precise per node information a fairness scheme is presented.

Flexibility of the transport solution means that each mode of transport can be used separately hence an application requiring only fair transport is not required to implement the whole transport solution. Proposed transport solution is flexible in terms that how an event is to be transported e.g., fair or real-time, can be set either prior to node deployment in the event definitions or upon event occurrence it can be decided by the actors.

The proposed transport layer is shown in Table 4.1. It is independent of the underlying routing and medium access layers. It is divided into two major parts as *sensors-to-actors* and *actor-to-sensors* transport according to the flow of information and due to the difference in their reliability syntax.

Table 4.1 Proposed transport layer for wireless sensor and actor networks.

Application	Application-specific event definition and transport mode selection	
Transport	Actor-to-sensors Transport	Sensors-to-actors transport SETM, FETM, PETM, TETM Congestion Control Scheme
Network	Minimum hop routing / Directed Diffusion / DSR	
Data Link	S-MAC / IEEE 802.11 / TDMA	
Physical	Radio - Industrial, scientific and medical (ISM) bands	

The sensors-to-actors transport is subdivided in different transport modes, simple event transport mode (SETM), fair event transport mode (FETM), prioritized event transport mode (PETM) and time-bound event transport mode (TETM).

4.2 Sensors-to-Actors Reliable Transport

The sensors-to-actor information flows, in WSANs are mainly compromised of event, periodic and request initiated flows. The later of the two flows, are not continuous as the nodes report their current sensor readings to the actors which are generally not more than few bytes. On the other hand, event information flow is continuous and a number of nodes report the event information to the actors, until event mitigates. Sensor networks are event driven networks, with their basic duty to deliver event information reliably to destination. Therefore, reliable transport of information between sensors and actors which is initiated by events is only considered in this study. In this study, the terms *upstream transport* is also used to refer to sensors-to-actors transport.

Sensor nodes are deployed to detect a number of application defined events. Each event has its own characteristics in terms of reporting rate, importance and reliability requirements. Hence, information transport from sensors-to-actors needs to achieve application defined goals for the event. It is shown in Table 1, that the proposed sensor-to-actors reliable transport has been further subdivided to support different transport modes. The need for these different transport modes in sensors-to-actors transport is justified in section 3.2. Congestion control is one feature that interacts with all transport modes in sensors-to-actors information flow. As, all event flows are prone to congestion due to sudden flow of information from large number of sensors to few actors.

4.2.1 Reliability in Sensors-to-Actors Transport

The goal of a transport protocol is to take necessary measures for ensuring reliable information transport between source and destination. Issues like flow control, error detection and recovery, congestion detection, avoidance and control are related to transport layer. Error detection and recovery for sensors-to-actors transport is not required in most WSANs applications due to the presence of redundant information. An occasional loss of packets is tolerable in sensors-to-actors flow, as described in section 2.3.

Reliability for upstream transport in existing literature has been defined in terms of number of packets received at the destination, as *application defined reliability* (Akan, & Akyildiz, 2005). Instead of application defined reliability, some previous works had aimed to provide maximum throughput of the system. In this study, the later is defined as *network based reliability*. The proposed sensors-to-actors transport is able to provide following reliability syntaxes:

- *Application defined reliability*: An application can specify that for reliable reception of event/non-event information, the destination (actor) must receive a certain amount of packets from the source(s) with in a specified duration of time. Therefore, reliability is measured in terms of total number of packets received by the destination to the total number of required packets (application defined); within a certain time period. The transport mechanism achieves the required level of reliability by increasing or decreasing the reporting rate of node and mitigating congestion. The objective behind application based reliability is that once reliability has been achieved further increase of reporting rate will not increase the application's knowledge about the event. Hence, further increase will result in wastage of energy in terms of unnecessary transmissions.
- *Network based reliability*: It is based on number of event reporting nodes and network conditions. The basic consideration is that events occur for small durations and maximum event related information must be transported to the actors. Hence, the nodes provide maximum throughput while avoiding or controlling congestion by adjusting their reporting rates according to network conditions.

In WSANs, upon the occurrence of event, sensor nodes send the event information to the actors according to a predefined reporting rate set by the application for each event. This reporting rate is defined as *initial reporting rate*. Since reliability is event based not node based, a maximum reporting rate can not be specified to nodes, prior to node deployment. In most of the cases, overall event information is important, not just the event information generated by a single node. Since the initial reporting rate is kept low in order to avoid congestion, the number of nodes which will detect the event is not known prior to event occurrence. As a result, reporting rate of nodes needs to be adjusted in order to achieve reliability. This adjustment can be made either by the destination (actors/sink) or the nodes themselves.

The advantage of destination based reporting rate control is that the destination knows the number of reporting the event. Hence, the destination can adjust the reporting rate of all the nodes. Rate adjustment signal can be periodically broadcasted to all nodes which are reporting the same event. For example, ESRT (Akan, & Akyildiz, 2005) uses a high frequency signal to achieve this goal. In application such as mining, the physical conditions (such as rocks) do not allow all nodes to directly receive these signals.

The sensors-to-destination flow is characterized by a number of individual flows. In destination based rate control, if a node in a single flow is congested then all the nodes in the network have to decrease their reporting rate in order to avoid congestion. This limits the overall throughput of the networks. As a result, achieving maximum throughput is very difficult in pure destination based solution. On the other hand, maximum throughput can be achieved by allowing nodes to adjust their reporting rates, without destination's interruption but will result in energy wastage; if the required reliability level is less than maximum throughput.

Since sensors-to-actors information flow is characterized by more than one unique flow, the proposed transport solution uses a flow based transport solution. It is shown in Figure 4.1 that during the transport of event information from sensor nodes, more than one first hop node (from the actor) is involved in information routing.

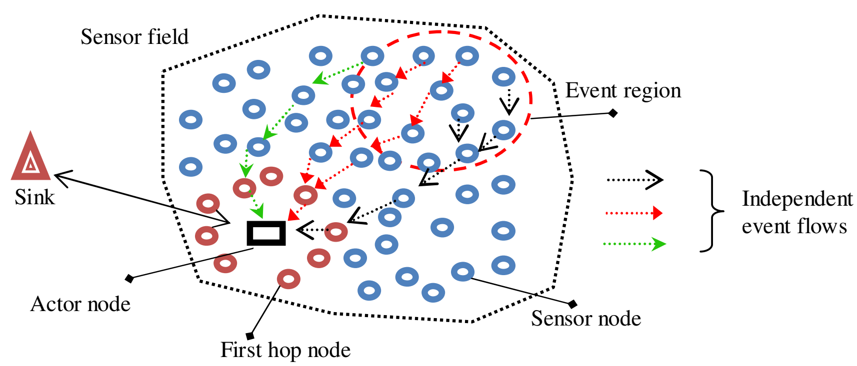

Figure 4.1 Multiple information flows on event occurrence from event region to actor.

The flow associated with each first hop node is a separate flow. On actor's instructions, the first hop node in proposed transport solution controls the reporting rate of the nodes in the flow. The advantages of using a flow based solution are listed below:

- Flow based rate control divides the sensors-to-actors event flow into multiple small flows. Therefore, congestion in a single flow does not affect the whole sensors-to-actors transport.
- For adjusting the reporting rate of nodes, the destination is not required to broadcast a high frequency signal for all the event reporting nodes. All packets received at the destination are from first hop nodes. Therefore, in flow based rate control, the destination adjusts the reporting rate of first hop nodes, which further adjusts the reporting rate of its flow members.
- Flow based rate control can also achieve maximum throughput by decoupling the control link between destination and first hop node. So each node depending on its local network's conditions adjusts the reporting rate to achieve maximum throughput.

4.2.2 Modes of Operation for Sensors-to-Actors Transport

Sensors-to-actors information transport is comprised of different transport modes. The selection of a transport mode can be *node based* where nodes are preprogrammed to report a certain event in a specific mode, or *actor based* where the actor at event detection can indicate nodes to report the event in some specific mode. The basic goals of these modes are as follows:

1. Simple Event Transport Mode (SETM): Sensor nodes report event to the actor in order to provide general event region information. Simple event reporting aims to achieve either application defined reliability or maximum system throughput irrespective of per node share at the destination. SETM is suitable for low priority events and also for obtaining a general sensing field status from the nodes.

2. Fair Event Transport Mode (FETM): This transport mode is responsible for providing same per node throughput at the actor. Sensor nodes within a single flow adjust their reporting rates in order to fairly distribute the bandwidth among all the event reporting nodes. FETM also aims to achieve either fair application defined reliability (i.e., application specified same per node throughput) or maximum throughput.
3. Prioritized Event Transport Mode (PETM): In sensor networks depending on application needs information regarding different events, node, or region can be transported with priority. In this study, event based priority for multiple events is presented which aims to distribute the system bandwidth among different event reporting nodes depending on their initial reporting rates. As a result, nodes with high reporting rates deliver more packets to the actor than nodes with lower reporting rates; irrespective of node distance from the actor. PETM for each reported event achieves either application defined reliability or provides maximum throughput.
4. Time-bound Event Transport Mode (TETM): This transport mode is responsible for delivering time-bound event packets to the destination within their respective deadlines. In TETM, application defined reliability is given in terms of number of in-time packets received to the required in-time packets specified by the application within a certain time period.

4.2.3 Congestion in Sensors-to-Actors Information Flow

The sensor-to-actors flow of information can be periodic, on request and on event occurrence. In case of periodic and on actor/sink request, the sensor node or a group of sensor nodes can transmit their sensor's readings to the destination. This kind of flow is generally for small duration of time (part of second) and comprises of just a few bytes (e.g., temperature or pressure). Periodic and on-request flows do not enforce a burden on the network, as the general purpose of these flows is to update the systems knowledge of the sensing field. In case of large scalable networks these periodic readings are alternatively obtained on region basis, in order to decrease the load on the network.

On event occurrence due to high node density, a large number of nodes from the event region continuously start sending event information to the actors. This sudden impulse of flow immediately increases the load on the network, resulting into congestion. Interference increases as nearby sensors suddenly start to transmit information at the same time. Packet drops occur due to the overall affect of congestion and interference. As a result, the destination does not get the correct picture of event region. Considerably high amount of energy loss is observed due to packet losses. In order to decrease congestion on event occurrence, the initial reporting rate of nodes is kept low.

Due to dense deployment of nodes and the nature of event, it is not possible to predict how many nodes or how much region will be affected on event occurrence. Selecting a low initial reporting rate can not guarantee that congestion will not occur immediately. The destination's view of the event region is also limited by keeping the initial reporting rate of nodes considerably low. Therefore, congestion avoidance and control is necessary for sensors-to-actors information transport. As a result, in the proposed transport solution (shown in figure 4.1) all the transport modes interact with the congestion control scheme.

4.2.4 Evaluation Metrics for Sensors-to-Actors Reliable Transport

The proposed transport solution is compared with different protocols that were presented for transport layer in existing literature on sensor networks. The evaluation metrics used for proposed sensors-to-actors transport are listed below:

- Application defined reliability: The purpose of the proposed transport solution is to achieve application defined reliability. It is defined as the required number of packets received per unit time for reliable detection of an event.
- Throughput: It is defined as number of packets received by the destination per second (or per unit time).

- Energy consumption: Energy consumption or residual energy of the network is used to measure the energy efficiency of the proposed modes of transport with existing works.
- Scalability: It is defined as the number of nodes in the network or the number of nodes reporting event information to a single network.
- Density: It is the number of nodes within the radio range of a single node. The greater the density of the nodes higher will be interference and congestion.
- Per node throughput: The number of packets received per second by the destination from each source node. This is the basic metric for fairness.
- Latency: The time in which application defined reliability is achieved. For time bound information transport, this defines the time each packet takes from the source to the destination.
- Multiple events: The transport of multiple events according to their importance, in terms of reporting rate.

4.3 Actor-to-Sensors Reliable Transport

The reliable actor-to-sensors transport is similar to one-to-many and downstream transport in wireless sensor networks. The syntax of reliability for downstream transport is guaranteed delivery of information. This kind of transport is generally used for the following purposes.

- To re-task a group of nodes for a different purpose; since nodes can perform different sensing functionalities
- To reprogram a group of nodes like loading new binary files
- For upgrading event database of sensor nodes
- To query the status of the network

WSANs use wireless communication mode for information propagation. The wireless channels in WSANS are prone to high channel error rates when compared with wireless channels in ad hoc networks. In the recent experimental studies (Son, Krishnamachari, & Heidemann, 2006; Zhao, & Govindan, 2003; Zhou, He, Stankovic, & Abdelzaher, 2005) it was shown that in WSANs, wireless link quality varies over space and time. As a result, packet drop ratio due to poor quality radio links is high. Since sensor nodes are battery-powered devices, it is important to conserve energy in sensor networks.

The goal of proposed actor-to-sensors transport is to transport information with high reliability and minimum energy expenditure. In this study, along with reliability and energy conservation, latency is also considered, which is defined as the time in which information is completely transported to all the destinations. In order to decrease latency especially in error prone wireless links, it is required that the nodes retransmit lost information quickly and more frequently. The application needs to define a tradeoff between latency and energy consumption. In this study, the proposed actor-to-sensors transport solution has defined different parameters for latency and energy consumption. It is left to application to choose the appropriate settings for these parameters, in order to, trade off between latency and energy consumption.

4.3.1 Evaluation Metrics for Actor-to-Sensors Reliable Transport

The proposed transport solution evaluates the actor-to-sensors information dissemination according to the flowing metrics:

- Reliability: It is defined as the 100% reception of all data packets transmitted by the actor to all the sensor nodes.
- Channel error rates: The proposed transport solution is aimed to provide reliability under different channel error rates. Channel error rate can be as high as 20% in closed environments in sensor networks (Wan, Campbell, & Krishnamurthy, 2005).
- Overhead: Number of extra transmissions required to transport the information from actor to sensors. Extra transmissions include: NACKS (negative acknowledgements), NACK replies and different control transmissions. required in actor-to-sensors transport.

- Latency: The time in which the complete data (file) from the actor is transported successfully to all the sensors.
- Scalability: Number of sensor nodes to which data in transported by a single actor.

4.4 Design Considerations for Proposed Transport Solution

A transport solution is proposed in this study that is independent of the underlying layers. The basic design considerations regarding the proposed transport solution and the architecture of WSAN are as follows:

1. The sensor and actor nodes which are used for environmental control and action purposes are stationery.
2. The topology of the WSAN is not fixed. Nodes are scattered or thrown in the medium where new nodes can be distributed at any time, according to application's need. Therefore, the topology of the network is random.
3. Since the nodes are scattered in the phenomenon, node density is not uniform in the network. The density of the nodes around the critical sensing areas is deemed to be higher than the others.
4. Nodes deployed in the network are position aware. Position of a node can be determined by using GPS (Global positioning system) (Wellenhof, Lichtenegger, & Collins, 1997) but for large scale sensor networks having cheap, small and energy constrained sensors; GPS is not a suitable solution (Bulusu, Heidemann, & Estrin, 2000). In the existing literature, a number of different schemes are presented for localization, in which nodes can find their relative position with respect to some special position aware nodes in the network (Fang, Du, & Ning, 2005; Hu, & Evans, 2004; Savarese, Rabaey, & Beutel, 2001).
5. The nodes are assumed to be synchronized. The techniques presented by Elson, & Romer, (2003), Sichitiu, Elson, Estrin, & Shenker, (2003) and Younis, & Fahmy, (2005) can be used for time synchronization among nodes.

6. The actors are power rich devices that are capable of directly communicating with each other or via actor-to-actor transmissions.
7. The sensor nodes operate on radio frequency of 916/433 MHz, with a radio range of 20-40 meters. Sensors communicate with the actors on the same frequency using hop-by-hop transmissions. Sensor nodes use fix transmission power and do not vary their power to achieve more fidelity. The transmission power, reception power, radio propagation model and other characteristics of sensor nodes mirror that of basic Mica Motes (Hill, & Culler, 2002).
8. Few packet drops can be ignored in sensors-to-actors transport due to the syntax of reliability in this transport (as discussed in section 4.2.1). Underlying routing protocols in coordination with the MAC layer can select the best path for the routing of data. The congestion that occurs due to lossy wireless links is not taken into account for the proposed sensor-to-actors transport.
9. In order to avoid packet drops due to interference in high node densities, the proposed transport solution has suggested a scheduling scheme at the transport layer. Removing interference is the duty of underlying MAC layer and the scheduling scheme only decreases interferences but does not eliminates it.
10. Bit errors or erroneous packets are not considered in the proposed transport solution.

4.5 Simulation Environment

Sensor networks face many problems that do not arise in other type of networks. Power constraints, limited hardware, low quality radio links, high density and higher number of nodes are some of these problems. The goal for any simulator is to model and predict the behavior of an algorithm or protocol considering real world environment.

The simulation environment for wireless sensor networks requires not only to perform specified simulations of algorithms but also needs to have models for wireless transmissions, battery models, extremely scalable, efficient for large simulations and should be open source in nature for modifications.

The performance evaluation of the proposed transport solution is done using network simulator, *NS-2* (Breslau & et al., 2000). It is an object-oriented, packet-level discrete event simulator for wired, wireless and satellite communication. Due to the intense use of *NS* by research industry, it can be considered as de facto standard for network simulations.

NS-2 is also an open source simulator, which allows the user to modify the implemented protocols according to user needs. The NS-2 implements radio propagation models, MAC protocols, interface queue, link layer and address resolution protocol model for wireless communication. NS-2 sensor simulation tool is a modification of NS's mobile ad hoc simulation tool and Downard, (2004), presented support for creating and triggering external phenomena's in sensor fields.

NS is an object oriented simulator, with an OTcl interpreter as a frontend. The simulator supports a class hierarchy in C++ and a similar class hierarchy within the OTcl interpreter. The two hierarchies are closely related to each other; from the user's perspective, there is a one-to-one correspondence between a class in the interpreted hierarchy and one in the compiled hierarchy. The root of this hierarchy is the class TclObject. Users create new simulator objects through the interpreter; these objects are instantiated within the interpreter, and are closely mirrored by a corresponding object in the compiled hierarchy.

A number of other simulation tools are available for performance evaluation of protocols in sensor networks. Following are the basic features of some well known simulators for sensor networks:

- *GloMoSim:* Global mobile information systems simulation library (GloMoSim) (Zeng, Bagrodia, & Gerla, 1998), is specific for mobile wireless networks and is built as a set of libraries which use Parsec (Parallel

Simulation Environment for Complex Systems); a C-based discrete event simulation language. It has a layered architecture with considerably easy plug-in capability. It is effective for IP network simulations but it is not capable of simulating any other type of network. This ensures that many sensor networks can not be accurately or fully simulated by GloMoSim.

- *OPNET:* (OPNET Modeler Wireless Suit, 2000) Like ns-2, OPNET is a discrete event based, object oriented and general purpose network simulator. Different sensor-specific hardware, radio propagation models and energy models can be simulated in OPNET. However, it does not currently support many protocols for sensor networks and is only available in commercial form.
- *SensorSim:* (Park, Savvides, & Srivastava, 2000) This is an extension of NS-*2* which provides battery models, radio propagation models, sensor channel models and a lightweight protocol stack for sensor networks. However, *SensorSim* is in its early stages with little documentation available, currently does not support a scalable network and is not available publicly.
- *J-Sim/Java-Sim:* (Sobeih, & et al., 2006) It is a modular java-based simulator for wired networks but later it was extended to support wireless network features. J-Sim is more scalable than NS-2 but supports little features for wireless communication (only 802.11 MAC layer support available). J-Sim is complicated to use, considerably slow and introduces unnecessary communication overheads in the inter-communication model.
- *SENSE:* (Sundresh, Kim, & Agha, 2004) It attempts to implement the same features and functionalities as that of NS-2. SENSE is faster and scalable than NS-2 but still it remains relatively obscured. The original implementation does not offer a great variety of protocols and requires significant background of sensor network protocols to start with.

4.6 Actor Selection Procedure for Proposed Transport Solution

The proposed transport solution deals with two opposite flow directions i.e., sensors-to-actors and actor-to-sensors. The common factor in both of these flows is the presence of multiple actors in the sensor field. Nodes either send or receive information from their neighboring actors. An actor selection procedure is required to select an appropriate actor for a node. Actor selection can be done on the basis of metrics like latency, efficiency, load etc. However, in this study a node selects the actor on the basis of minimum hop distance. As a result, minimum number of transmissions will be required for sending or receiving information, resulting into energy saving. The nodes sending/receiving information from a specific actor are termed as *members* of the actor.

During the network configuration, actors broadcast *presence beacon* to their neighboring nodes. The beacon contains actor's address, hop count and a time-stamp. The beacon is broadcasted in the network using controlled flooding. Nodes after receiving the first beacon packet become the members of the actor, from which the packet has been received. After incrementing the packet, the nodes wait for a small random amount of time before further broadcasting this packet to their neighbor nodes. If during this time nodes receive the same packet more than three times then they cancel their broadcast. This ensures controlled flooding and maximum coverage. Wan, Campbell, & Krishnamurthy, (2005) observes that in broadcast environment broadcasting the same packet more than three times gives either no or very less coverage.

After receiving a packet with a hop count greater or equal to the already accepted actor, member nodes will discard the packet. If member nodes receive a packet with hop count smaller than the one they have previously accepted, then they will discard the previous membership and will make a new membership (even if it from the same actor). Moreover, they will further broadcast this packet after incrementing the hop count. The resultant of actor selection procedure is shown in Figure 4.2, with 100 sensor nodes and 4 actor nodes in a 100x100m sensor field.

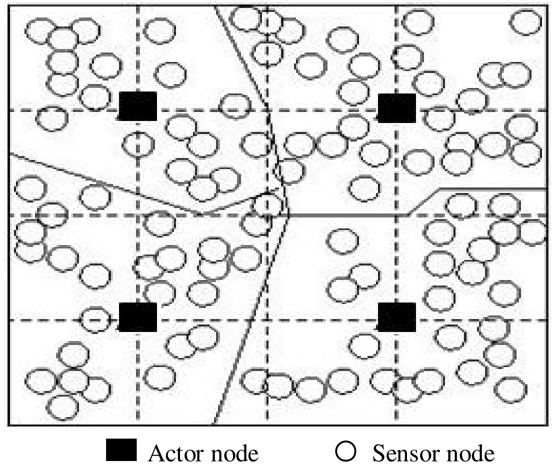

Figure 4.2 Membership of nodes after actor selection procedure, in 100x100m sensor field with 100 sensor nodes and 4 actor nodes.

CHAPTER FIVE
A CONGESTION CONTROL SCHEME FOR WIRELESS SENSOR AND ACTOR NETWORKS

5.1 Motivation

The sensors-to-actors or upstream information flow, results into congestion in Wireless Sensor and Actor Networks (WSANs). Congestion is considered as the basic source of packet drops in the proposed sensors-to-actors transport.

Information in WSANs is transported from source to destination using multi-hop forwarding. A forwarding/routing node after receiving the packet temporarily buffers the packet before forwarding it to the next hop node. If the incoming rate of packets at a node is greater than its outgoing rate then this results in buffer overflow and congestion at the node. In WSANs, this situation occurs on event occurrence as sudden impulse of information flows from many sources to a single or few destinations.

When fixed transmission power of nodes is considered, with increasing the number of event reporting nodes greater will be the degree of congestion. An important factor in the design of a congestion control scheme is the density of the network. Since nodes are scattered in the sensing field, the node arrangement is random. As a result, the density of the network will be greater in some regions while less in others. Regions where events may occur more frequently (e.g., where miners are working in the mining application) the density of the network will be high. Moscibroda, (2007), shows that, due to random deployment, the difference between densities in the network will be even more extreme in "worst-case" networks i.e., arbitrarily deployed sensor networks.

With the increase in the number of nodes within the event region, packets will be dropped due to collisions at the MAC layer. Also, contention for medium access increases as packets do not get a chance for transmission in busy medium. This results into an increase in buffer size of nodes in dense networks. This fact is demonstrated in Figure 5.1, where the average buffer occupancy of event reporting nodes under variable densities is shown.

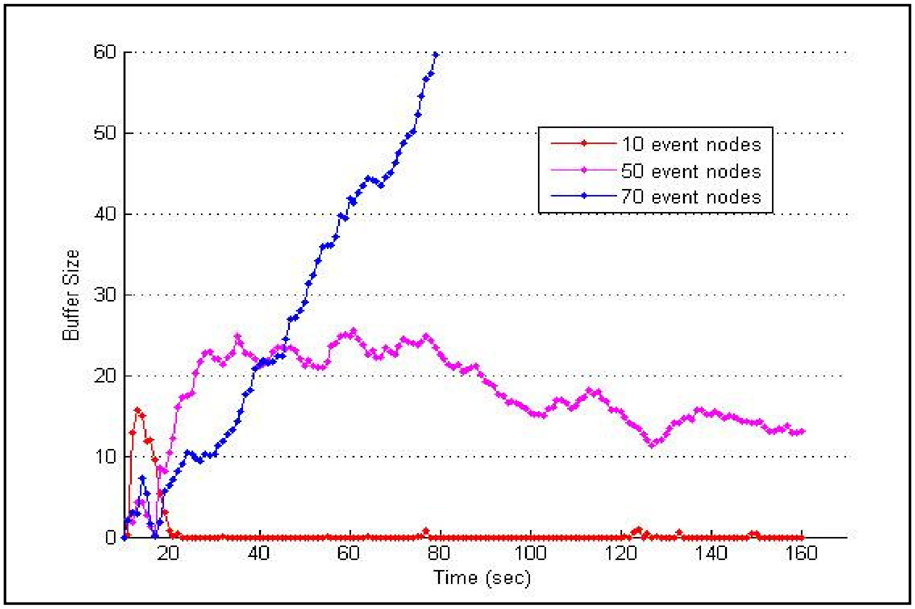

Figure 5.1 Average buffer size of 10, 50 and 70 event reporting nodes using minimum hop packet forwarding in a 100x100m sensor field.

The network comprises of 100 sensor nodes, which are randomly deployed in 100x100m. The event reporting region is centered at (40,40) coordinates and having a radius of 20m. 10, 50 and 70 event nodes, report event to maintain a constant reporting rate (70 packets per second) at the actor; placed at (90,90) coordinates. The event nodes report event to the actor using minimum hop forwarding without any congestion control scheme. It is evident from Figure 5.1, that as the number of event reporting nodes increase within the event reporting region, the average buffer occupancy of event reporting nodes also increases.

A number of studies (Hu, & et al., 2005; Rangwala et al., 2006; Shigang & Na, (2006); Wang, & et al., 2007) had proposed different techniques for solving the issue of congestion in sensor networks either by rate control or by buffer management. These techniques use different metrics like buffer size, packet inter-arrival time, packet service time, channel sampling and traffic load assessment to detect congestion. Density of the network is an important issue, which is not considered as a primary design issue in these techniques. The existing congestion detection techniques when used alone are unable to efficiently tackle congestion in dense networks. In high node densities, metrics like buffer occupancy, packet arrival time or service time increases due to collisions and contention of packets.

This study has addressed the issue of congestion control, in order to ensure maximum throughput of the system in dense networks. The proposed congestion control scheme uses packet delivery time and buffer size as the basic metrics for congestion detection. A congestion mitigation scheme not only needs to detect congestion efficiently but also needs to adjust the reporting rate of nodes in order to avoid or remove congestion. Rate-control mechanism is an important and implicit part of congestion control protocols. In the existing literature, for adjusting the reporting rates of nodes either a destination-based (Akan & Akyildiz, 2005) or a hop-by-hop (in-network) solution is used (Wan, & et al., 2003; Wang, Li, Sohraby, Daneshmand, & Hu, 2007).

In the destination-based solution, the destination is responsible for rate control while in in-network solution nodes adjust their reporting rates according to the local congestion status. In both cases, event reporting nodes, report event at a rate which is in accordance with (local/network) congestion status, using jittered forwarding of packets at the transport layer. Destination-based rate control schemes are slow to react to congestion as the congestion information needs to be transported to the destination which then adjusts the reporting rate of nodes.

The proposed scheme uses in-network congestion mitigation technique in which nodes adjust the reporting rates to avoid and mitigate congestion. Also, an important observation made in this study is that by using a schedule-based scheme at the transport layer for forwarding packets helps to avoids packet collisions and increases the packet delivery ratio even in high densities as compared to jittered forwarding. Hence, this work introduces a TDMA-like (schedule-based) scheme at the transport layer for orderly forwarding packets to the underlying layers which is highly effective in high densities.

5.2 Network Model

The basic system related definitions, network setup and assumptions for the proposed congestion control scheme are presented in this section. The network comprises of non mobile wireless sensor nodes and a sink. The nodes in the network are categorized as event reporting (*E-REP*), routing (*E-R*), reporting and routing (*E-REP-R*), and idle nodes. If *b,c,d* are the nodes routing through node *e*, then *b,c,d* are the previous hop or child nodes of *e* and *e* is the next hop node of *b,c,d*, as shown in Figure 5.2. All nodes routing event information through node *e* are associated with the same information flow.

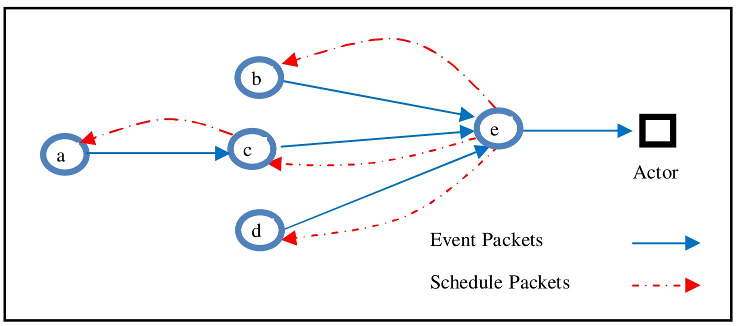

Figure 5.2 Flow of event and schedule packets.

All nodes which are directly capable of hearing the transmissions of a node are its neighbor nodes. In other word, the number of nodes with in the radio range of a single node *x*, are the neighbors of *x*. A node in the proposed sensors-to-actors transport requires its neighboring node information. Each node maintains a *next hop table* which contains the list of its possible next hop nodes, which are at minimum hop distance from the actor. Different protocols require neighbor node information for variable purposes like energy status, current mode of node, congestion status, and for routing purposes etc. This kind of information can also be obtained from the underlying layers. The proposed transport layer considers no assistance from the underlying layers in terms of cross layer interaction. The next hop table is established in the proposed transport solution, during the initial network setup using the presence beacon broadcast (discussed in section 4.6). During event reporting, nodes randomly select a single next hop node from their table and forward their packets through that node.

Nodes maintain successive fixed size data (γ) and schedule (δ) intervals throughout their life time. During the data intervals, nodes route *available event information* and during schedule intervals, routing nodes send transmission schedule for their previous hop nodes. The available event information at a node can contain, both event packets generated by the node itself and packets received from previous hop nodes. The schedule comprises of slot length (λ sec), total number of slots and allocated number of slots for a previous hop node. Slot length is defined as a *time duration during which a node forwards a single packet*.

Assuming that nodes *a, b, c, d* in Figure 5.2 are *E-REP* nodes and are reporting the same event. Nodes *b, c, d* are at two hop distance from the actor and node *e* is their next hop node which is routing the event information. Node *a* is at three hop distance from the actor with node *c* as its next hop node. Node *e* will generate a schedule for its pervious hop nodes which is shown in Table 5.1, (details of slot length calculation and slot allocation will be explained in next section).

Table 5.1 Transmission schedule generated by node *e*.

Node ID	Total Slots	Initial Slot	End Slot	Slot length
B	4	1	1	.1
C	4	2	3	.1
D	4	4	4	.1

Nodes *b, c* and *d* will divide their data interval into 0.1 sec (initial slot length on event occurrence) intervals and will forward one event packet to node *e* during their allocated slots. Furthermore, node *c* will generate and forward a schedule for node *a*.

The first hop node from the actor in the proposed congestion control scheme is responsible for sending the initial schedule to their previous hop nodes during the start of each schedule interval. The slot length for each data interval is updated by *E-*

REP-R and *E-R* node during schedule interval depending on their local network conditions. The slot length received from a parent node or next hop node is termed as *basic slot length* (*BSL*) while the slot length calculated by the node itself is termed as *local slot length* (*LSL*). *E-REP-R* and *E-R* nodes after receiving the schedule from parent node either sends *LSL* or *BSL* to their previous hop nodes in the schedule packet (see section 5.3.3.1).

Each node follows the schedule received from their next hop node. In case if a node does not receive a new schedule during schedule interval, then during the next data interval it continues transmission by using the old schedule.

Slot length determines the reporting rate of a node during an interval. If the slot length is short, more traffic will be forwarded by the node and vice versa. TDMA (Time Division Multiple Access) uses fixed time slots that are assigned to sources. However, in the proposed schedule-based scheme, time slots are dynamically assigned during the schedule intervals depending on the average packet delivery time observed by nodes and their buffer size. TDMA requires strict synchronization among the nodes sharing a schedule. On the other hand, in proposed scheduling, synchronization is done between a node and its previous hop nodes. The previous hop node in turn generates a schedule for its previous hop node. Hence, proposed scheduling scheme does not require a network wide synchronization.

Packet delivery time for proposed congestion control scheme is defined as *the time a packet takes to reach from the transport buffer of a previous hop node to the next hop node's transport buffer*. Nodes maintain a queue at the transport layer and forward a single packet from it during their allocated slots. Packet delivery time not only includes service time but also the transmission time plus the reception time at the destination. Congestion is likely to occur if the packet delivery time of nodes exceed the packet delivery time of their previous hop node resulting into buffer overflow.

The proposed congestion control scheme divides the buffer size of nodes into three ranges low, medium (optimal), and high. The goal of this scheme is to maintain the buffer size of nodes within the optimal range. If the buffer size is below the optimal range, nodes will decrease the slot length for the next interval. Likewise, if the buffer size is above optimal range slot length is increased, to avoid any possible congestion and to achieve optimal range. Hence, each node adjusts the reporting rate (through slot length) of its previous hop nodes, so that the buffer is optimally utilized and to avoid congestion while providing high throughput.

5.3 Operation of Proposed Congestion Control Scheme

The operation of proposed congestion control scheme includes mechanism to detect congestion and to adjust the reporting rate (slot length) of nodes. In order to detect and remove congestion, the slot lengths are calculated by nodes depending on their local network conditions; in terms of average packet delivery time and buffer size. Other important operations of congestion control scheme consist of slot allocation to nodes on event occurrence and the general working of schedule based technique.

5.3.1 Slot Length Calculation

Nodes observe the average packet delivery time of their previous hop nodes in a data interval from the received packets. The average packet delivery time observed during the data interval is used as slot length for the next data interval. If a node's buffer is either under/over utilized during the data interval, then the node adjusts its slot length in order to optimally utilize the buffer.

Let λ_i^t be the slot length for the t^{th} interval; B_i^t and B_i^{t-1} be the buffer size for the t^{th} and $(t-1)^{th}$ interval of i^{th} node. Then in order to calculate appropriate slot length for the next interval λ_i^{t+1}, a node measures change in buffer occupancy (φB) and predicted buffer occupancy (ρB) for the next interval; similar to ESRT (Akan, & Akyildiz, 2005) as:

$$\varphi B_i^{t-1,t} = B_i^t - B_i^{t-1} \qquad (1)$$

$$\rho B_i^{t+1} = B_i^t + \varphi B_i^{t-1,t} \qquad (2)$$

In case the predicted buffer occupancy is not in the optimal buffer occupancy ($oB_{\min} \leftrightarrow oB_{\max}$) range, then nodes adjust their slot length for the next interval by adding or subtracting a deviation factor (ω) in the current slot length. The deviation factor ω^t for the i^{th} node, at the end of t^{th} interval is calculated as:

$$\text{Deviation}\ (\omega^t) = (\pm\,(oB\ - \rho B_i^{t+1})\ /\ oB\ \ )\ *\ \lambda_i^t \quad \text{where } oB\ = (oB_{\min} + oB_{\max})\ /\ 2 \qquad (3)$$

Hence, the slot length for the next interval will be:

$$\lambda_i^{t+1} = \lambda_i^t - \omega^t \quad (\text{if } \rho B_i^{t+1} < oB_{\min})$$

$$\lambda_i^{t+1} = \lambda_i^t + \omega^t \quad (\text{if } \rho B_i^{t+1} > oB_{\max})$$

$$\lambda_i^{t+1} = \lambda_i^t \quad (\text{if } \rho B_i^{t+1} \in (oB_{\min} \leftrightarrow oB_{\max}))$$

A zero buffer occupancy during an interval means that the link is under utilized and the packet forwarding rate of previous hop node is low. If the buffer occupancy is zero, then slot length for the next interval is decreased to half.

The pseudo code of slot calculation procedure is given below:

```
Slot Length Calculation Procedure, it is called at the start
of each schedule interval for routing and reporting & routing
nodes

1.  // AVERAGE_DELIVERY_DELAY observed during the last data
   interval SLOT_LENGTH_CURRENT (λ_i^t) = AVERAGE_DELIVERY_DELAY

2. // Initial case for the first time when event is detected
   and reported
        IF CURRENT_BUFF_SIZE = 0 AND AVERAGE_DELIVERY_DELAY =
0
           NEXT_SLOT_LENGTH (λ_i^{t+1}) = DEFAULT (0.1sec) // Slot
length for next interval
               GO TO STEP 7

3. IF CURRENT_BUFF_SIZE = 0 //Special case when reporting rate
   is low and no congestion
       λ_i^{t+1} = λ_i^t / 2
       GO TO STEP 7

4. Calculate PREDICTED_BUFF_SIZE (ρB) // For next interval
   using Eq. 1 & 2

5. IF (ρB < 0) // Special case if BUFF_SIZE for previous
   interval is less than the current interval
           pB = CURRENT_BUFF_SIZE

6.     IF (ρB < oB_min)
           Calculate DEVIATION FACTOR (ω^t) // using Eq. 3
           λ_i^{t+1} = λ_i^t - ω^t
       Else If (ρB > oB_max)
           Calculate DEVIATION FACTOR ω^t // using Eq. 3
            λ_i^{t+1} = λ_i^t + ω^t
       Else
             λ_i^{t+1} = λ_i^t

7.  TRANSMIT SCHEDULE
```

5.3.2 Slot Allocation

Nodes report event by indicating event's initial reporting rate (application defined for a particular event) to one of its next hop node. After initial event request, each *E-R* and *E-REP-R* node calculates a schedule at the start of schedule interval that comprises of slot length, total slots and allocated slots based on the number of nodes traversing through that node.

If any event E_1 has an application defined initial reporting rate R_1, which is the minimum reporting rate among all the events observed by the network. Then in order to simplify slot assignment, the reporting rates of other events are considered to be a factor of R_1; expressed as $2^n R_1$ while $n \geq 0$. The number of slots S_i^j assigned to j^{th} previous hop neighbor of the i^{th} node can be calculated based on the reporting rate R_j of the j^{th} node and the minimum reporting $R_i^{\min}$ traversing through the i^{th} node as: $R_j / R_i^{\min}$; where $R_j, R_i^{\min} \in 2^n R_1$ and $R_j \geq R_i^{\min}$. If k are the total number of nodes to be routed through the i^{th} node then it calculates total number of slots as $\sum_{j=0}^{k} R_j / R_i^{\min}$.

5.3.3 Operation of Schedule Based Scheme

The operation of schedule based scheme that is controlled by the first hop node (from the actor) is presented in this section. Each node maintains successive data and schedule intervals therefore the operation of schedule scheme can be subdivided into schedule interval and data interval operation.

5.3.3.1 Schedule Interval

During every schedule interval, nodes starting from the first hop nodes send their schedule packets to their previous hop nodes. Each previous hop node compares its local slot length (*LSL*) with the next hop node's basic slot length (*BSL*), before forwarding their schedule to their child nodes. There are three possibilities: local slot length less than, greater than or equal to basic slot length.

1. *LSL < BSL:* In this case the node receiving the schedule is locally less congested than its next hop node. Sending *LSL* to previous hop nodes instead of *BSL* allows previous hop nodes to send packets at a higher rate. But this

situation can result into congestion at the current node, as it is sending packets to next hop node at lower rate (i.e., at next hop node's transmitted *BSL*). To avoid any possible congestion, *BSL* is transmitted to next hop nodes. However, higher throughput can be received by sending *LSL* at the cost of few possible packet drops.

2. LSL > BSL: In this case the node receiving the schedule is more congested than its next hop node. The node will send *LSL* to their child nodes in the schedule. This allows the node to mitigate local congestion by requesting lesser packets from child nodes.
3. LSL ≈ BSL: In this situation, local and basic slot lengths are approximately equal, so the nodes will send basic slot length to their child nodes.

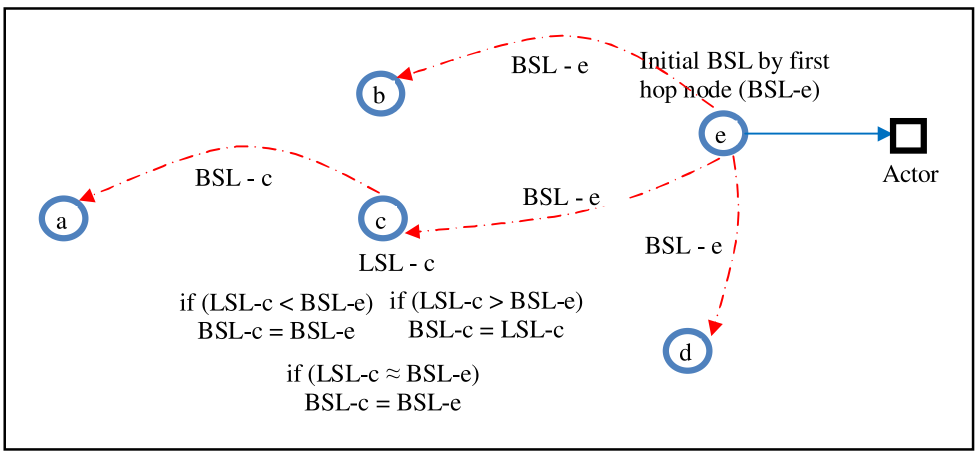

Figure 5.3 Flow of schedule packets and the selection of basic slot length and local slot length by intermediate nodes.

The flow of schedule packets are initiated from the first hop *E-R* node (*e*) and the slot length selection is done by the intermediate E-REP-R node (*c*), as shown in Figure 5.3. An actor (destination) is not involved in rate control mechanism. The schedule (*BSL*) transmitted from a node is only for its previous hop nodes. The previous hop node depending on its network condition can send a different schedule (*LSL*) to its child node. Nodes within a single flow can be transmitting information at different reporting rates in order to take advantage of congestion free links while limiting information flow at congested links. This allows proposed congestion control scheme to avoid and mitigate congestion while providing maximum throughput.

5.3.3.2 Data Interval

Each node maintains intervals equal to their selected slot length during the data intervals. At the expiry of each slot interval, a node checks whether it is allowed to send packet during the interval according to its schedule. An event reporting node (*E-REP*), during the allocated slots, sends an event packet to the next hop node, while routing nodes (*E-R*) simply forward a packet from the transport buffer to its next hop node in its allocated slot. Event reporting & routing nodes (*E-REP-R*) send new event packets as well as forward event packets from the transport queue according to their schedule.

Schedule packets are an implicit overhead in the proposed congestion control scheme. However, the numbers of schedule packets are very less and constant as compared to data packets. Let N be the total number of non-idle nodes in a single flow containing K number of event reporting nodes. Non-idle nodes include all *E-REP*, *E-R* and *E-REP-R* nodes. In this case $N-1$ will be the number of schedule packets generated during each schedule interval. If λ is the slot length during a data interval set by the first hop node of the flow, then $(1/ \lambda) * \gamma$ will be the number of data packets delivered to actor by the flow.

Smaller the value of λ, greater will be the number of packets delivered to the actor during data interval. λ is adjusted by nodes therefore its value is dependent on packet delivery time and buffer size of nodes. However, length of data interval (γ) is fixed. Longer the length of data interval smaller will be the overhead of schedule packets. Since slot length is adjusted in schedule interval of nodes, it will take more time to adjust the reporting rate of nodes and to achieve maximum throughput. Moreover, in case of congestion occurring during data interval, more packets will be dropped. On the other hand, smaller the length of data interval, greater will be the overhead of schedule packets and random will be the behavior of throughput, as reporting rate is adjusted frequently.

5.4 Simulation Results

The performance of proposed congestion control scheme is observed using network simulator NS-2, which is a scalable discrete-event simulator. The simulation scenario presents a wireless sensor network that consists of 100 sensor nodes randomly deployed in a 100 x 100 m field. Minimum hop packet forwarding is used at the routing layer. Length of data interval is 4 second while schedule interval is 1 second and the actors observe 10 second intervals. The configuration parameters for the simulations are summarized and given in Table 5.2.

Table 5.2 Simulation parameters for proposed congestion control.

Transport Layer	Proposed congestion control scheme
Network Layer	Minimum hop routing
MAC Layer	802.11
Deployment	Random
Packet length	36 bytes
IFQ Length	65 Packets
Transmit Power	0.660 W (fixed)
Receive Power	0.395 W
Radio Range	20m
Data interval	4 sec
Schedule interval	1 sec
Actor interval	10 sec

The performance of proposed congestion control scheme is evaluated in terms of packet delivery ratio, energy consumption and throughput observed at the actor. In order to verify the advantages of proposed scheduling scheme especially in dense networks, this study compares the efficiency of proposed congestion control scheme using both schedule-based (TDMA-like) forwarding and jittered forwarding at the transport layer.

5.4.1 Packet Delivery Ratio

Packet delivery/reception ratio decreases either by increasing the reporting rate of nodes or by increasing the density of the event reporting nodes (considering fixed transmission power for nodes). The simulation environment used to evaluate packet delivery ratio includes an event region, which is centered at coordinates (40, 40) and has a radius of 20 meters. The approximate density of event nodes within the event region is 10, 25 and 35 nodes for 20, 50 and 70 event reporting nodes respectively. In addition to event reporting nodes, 100 non-event nodes are in the network. In the simulation environment, a single actor at coordinates (90, 90) is used as the destination.

A sample node arrangement of 100 sensor nodes, in a 100x100 sensor field, with a single actor at coordinates (90, 90) is shown in Figure 5.4. The event area is shown with dotted lines and is centered at coordinates (40, 40). The event region contains 20 event reporting nodes.

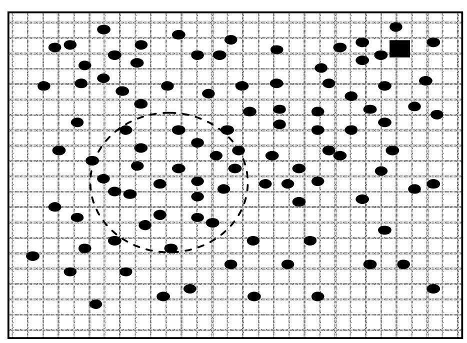

Figure 5.4 Arrangement of 100 nodes and an actor in a 100x100m sensor field.

In Figures 5.5, 5.6 and 5.7, the packet delivery ratio of 20, 50 and 70 event reporting nodes respectively are shown. For these event nodes, different simulations are carried out at per node reporting rate of 10, 20, 30, 40 and 50 packets per 10 sec interval. In all these simulations reporting rate of nodes is predefined and remains constant during event reporting. Packets are forwarded to the destination using minimum hop routing but no congestion control protocol is used at transport layer.

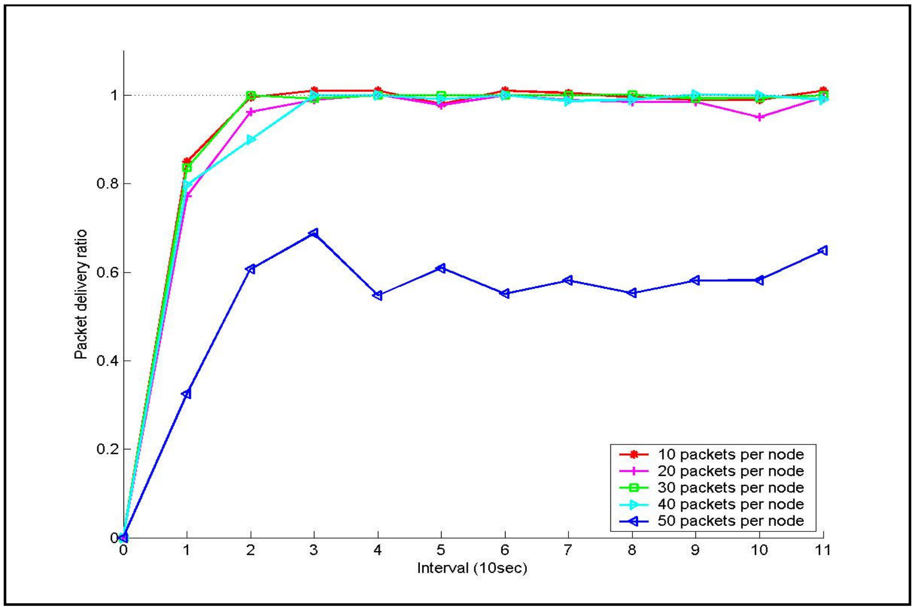

Figure 5.5 Packet delivery ratio of 20 event reporting nodes at different reporting rates using minimum hop routing and without congestion control.

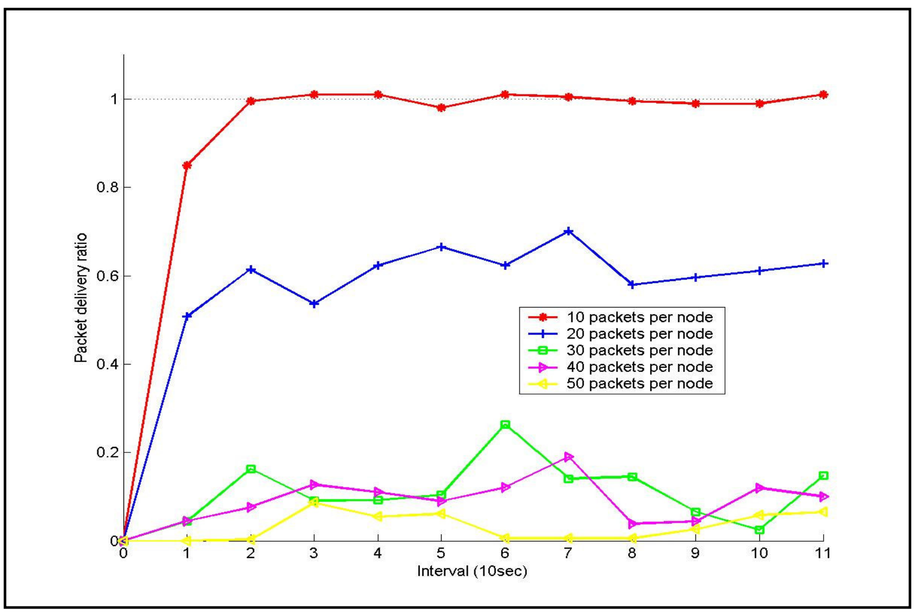

Figure 5.6 Packet delivery ratio of 50 event reporting nodes at different reporting rates using minimum hop routing and without congestion control.

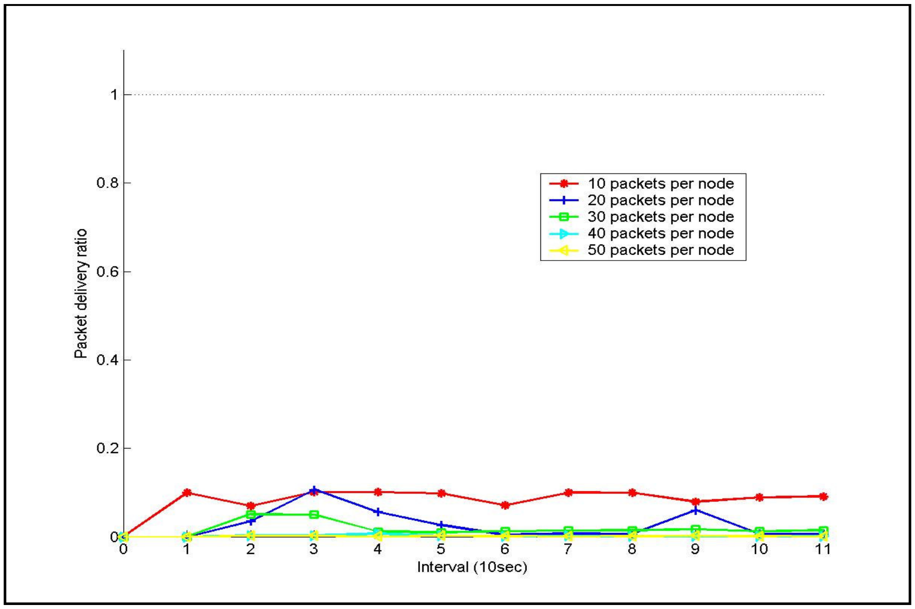

Figure 5.7 Packet receive ratio of 70 event reporting nodes at different reporting rates using minimum hop routing and without congestion control.

The results obtained from the simulations shown in Figures 5.5, 5.6 and 5.7, not only points to the necessity of a congestion control and rate control schemes but also indicate that using a fixed and predefined reporting rate for event reporting is not a viable solution in WSANs. The number of event reporting nodes is not known prior to event occurrence, if high numbers of nodes are reporting the event, even a small reporting rate can result into congestion. This is evident from Figure 5.7 in which packet delivery ratio of 70 event reporting nodes is shown, at very low reporting rate of 1 packet per second, the delivery ratio is well below 20%.

Another important factor is the density of the nodes in the event region. Since the density of event reporting nodes is less, congestion occurs at high reporting rates (Figure 5.5). Congestion occurs at very low reporting rates indicating that increase in number of nodes (density) affects the packet delivery ratio (Figure 5.6 and 5.7).

Source-based congestion control protocols like CODA (Wan, & et al., 2003), SenTCP (Hu, & et al., 2005) and (PCCP) (Wang, & et al., 2007) send congestion signal from congested node to source nodes. Intermediate nodes between the congested and source nodes, depending on their local network conditions further send this congestion to source nodes.

A pure source-based scheme for congestion removal which detects congestion on the basis of rate of change in buffer occupancy is implemented. In the implementation of source-based scheme, if a node's buffer is congested, then it sends congestion signal to source nodes to decrease their reporting rate. For controlling the reporting rate of nodes, AIMD (Additive Increase Multiplicative Decrease) mechanism is used.

The increment factor is kept relatively small in order to decrease the packet drops in case of congestion. Also, the decrement factor is kept high to immediately decrease the reporting rate of nodes and remove congestion. The performance of implemented source-based technique for congestion removal in sensor networks is shown in Figure 5.8.

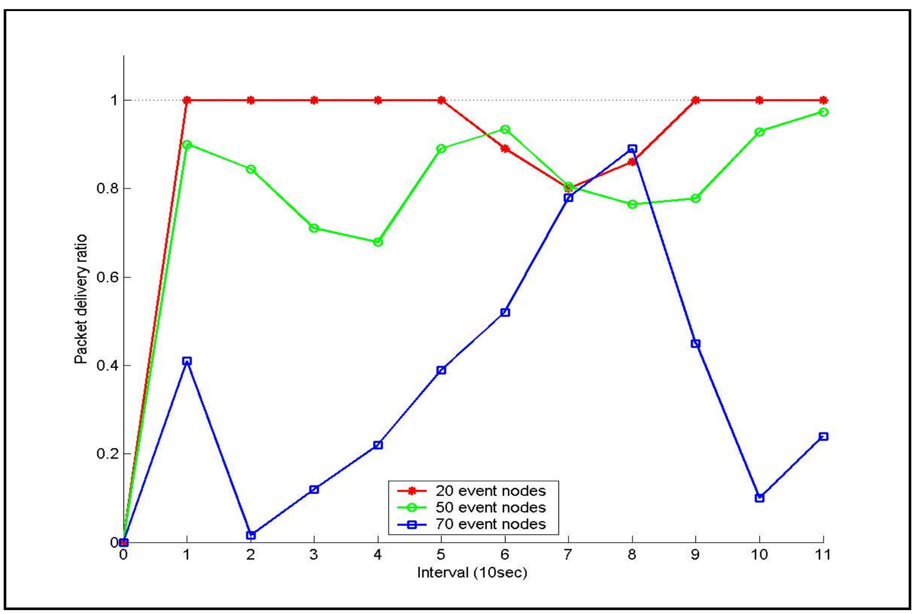

Figure 5.8 Packet delivery ratio of 20, 50, 70 reporting nodes using source-based congestion control scheme.

The initial reporting rate of event nodes is 1 packet per second. Nodes adjust their reporting rates periodically after every 10 second and an increment factor of 1.3 and decrement factor of 2 is used for the rate control. For 20 event reporting nodes, the packet delivery ratio initially is high because of the low reporting rate and fewer event reporting nodes (Figure 5.8). In case of congestion, the packet delivery ratio decreases but decreasing the reporting rate to half helps remove congestion.

AIMD based rate control schemes are not able to properly adjust the reporting rate of nodes as an increment/decrement factor is independent of the number of event reporting nodes. For 50 event nodes (Figure 5.8), the initial packet delivery ratio is high, but congestion occurs, as the reporting rate is increased and the overall reporting rate of the nodes increases beyond system's maximum achievable throughput.

During congestion, sending congestion signal to source nodes which are at multiple hops from congestion region is difficult. For 70 event reporting nodes, congestion occurs initially on event occurrence due to interference and busy channel as the density of the event nodes is considerably high (Figure 5.8).

The performance of proposed congestion control scheme using different number of event reporting nodes is shown in Figure 5.9.

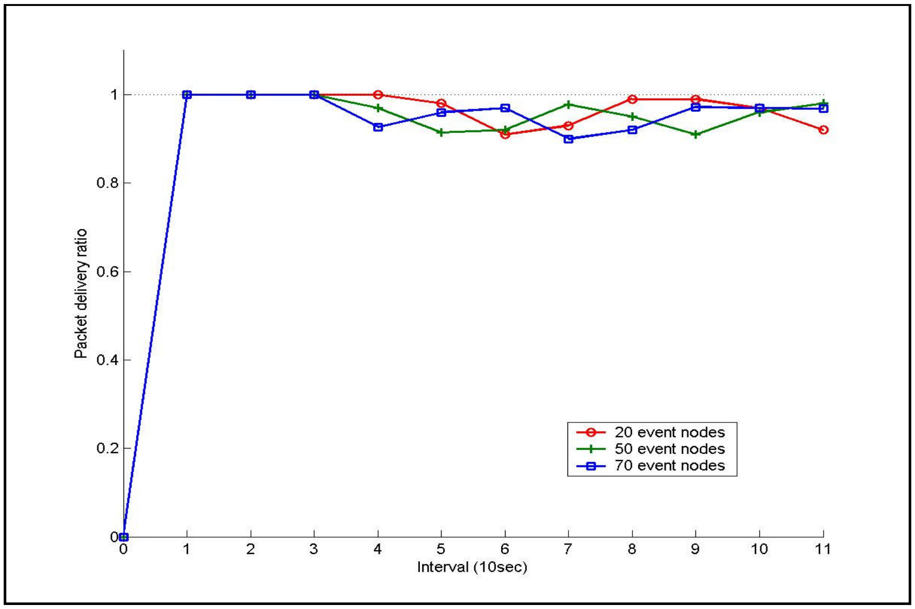

Figure 5.9 Packet delivery ratio of 20, 50, 70 event reporting nodes using proposed congestion control scheme.

The nodes aim to provide maximum throughput without congestion by optimally utilizing their buffers. The reporting rate of nodes is continuously adjusted in each schedule interval. Since the reporting rate can exceed the maximum supported rate of the link, or due to interference, few packets can be dropped at some links. This justifies the wavy behavior of the congestion control scheme in Figure 5.9.

However, the packet delivery ratio is above 90% for different number of event reporting nodes. This shows that the proposed congestion control scheme not only avoids but controls congestion by efficiently adjusting the reporting rate of congested nodes resulting into few packet drops.

5.4.2 Schedule Based vs. Jitter Based Packet Forwarding

The benefit of using an upper layer (transport) solution for reducing collisions in high densities is evaluated by increasing the number of nodes in the event region. The comparison of the proposed schedule based scheme with a simple jitter based forwarding scheme when both applied at transport layer, is shown in Figure 5.10.

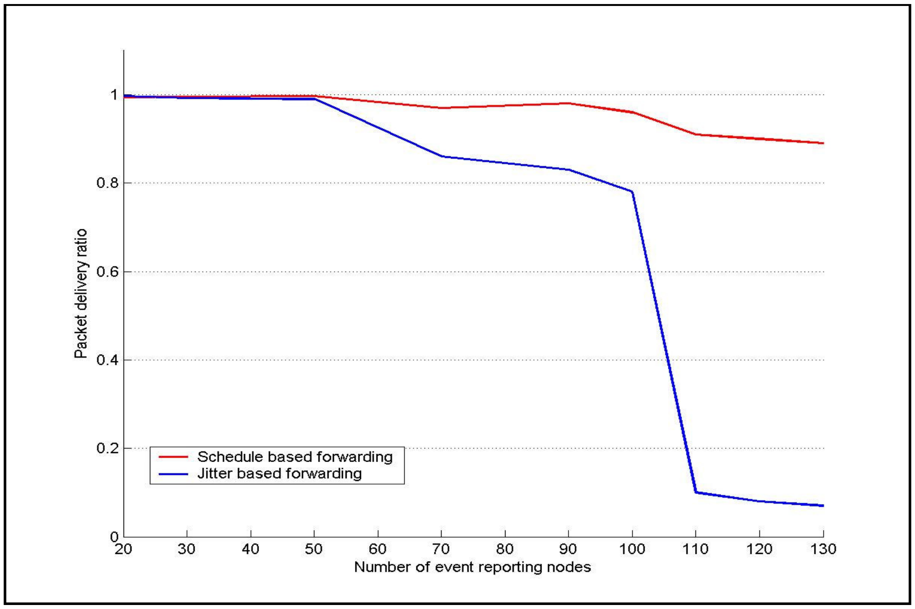

Figure 10 Packet delivery ratio of proposed congestion control scheme with schedule based and jitter based packet forwarding.

Proposed congestion control mechanism based on packet delivery time and buffer size is used in both the schemes, shown in Figure 5.10. However, in non-schedule version, routing nodes send reporting rate to their previous hop node instead of a schedule. The previous hop node uses jitter based forwarding at transport layer to avoid collisions.

Separate simulations are conducted for different densities and each simulation is run for 150sec. The density of event region is approximately half the number of event reporting nodes. The simulation settings used in the simulations are same as that described in section 5.4.1.

Packet delivery ratio drops in case of jitter based forwarding under high densities due to collisions, busy medium and buffer overflows. However, by using scheduled transmissions, packet delivery ratio increases (above 90%) resulting into better throughput of the system (Figure 5.10).

5.4.3 Throughput

Throughput of proposed congestion control scheme with scheduled and non-scheduled (jitter) forwarding at transport layer is shown in Figure 5.11. Also, proposed scheme is compared with source-based congestion control mechanism that uses AIMD rate control technique (as described in section 5.4.1).

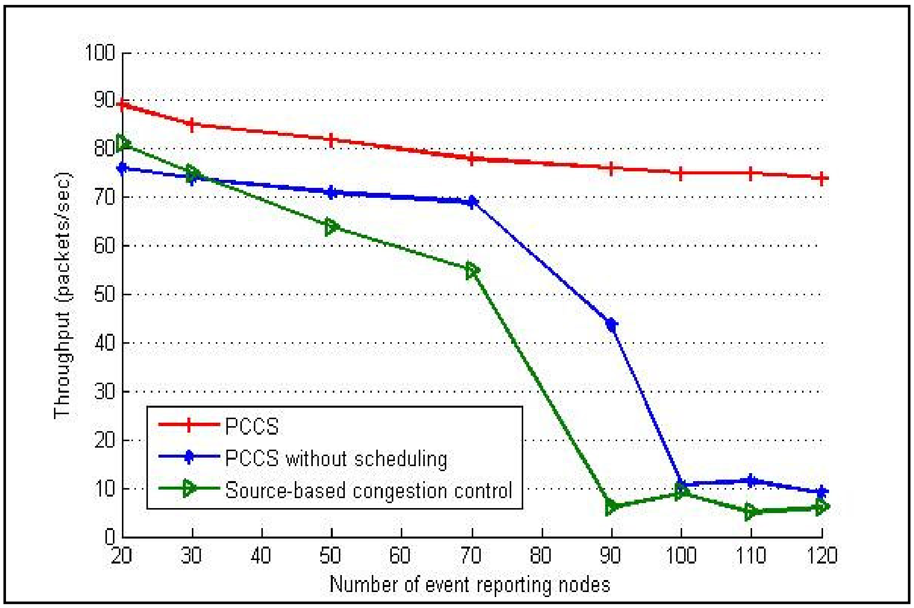

Figure 5.11 Throughput observed using Proposed Congestion Control Scheme (PCCS) and source based congestion control mechanism.

Separate simulations are conducted to obtain throughput in terms of packets per second at the destination (actor), observed for different number of event reporting nodes; during 150 second of event reporting (Figure 5.11). The simulation environment used to evaluate packet delivery ratio includes an event region which is centered at coordinates (40, 40) and has a radius of 20 meters. All the nodes report event through a single first hop node.

The throughput of source-based congestion control schemes, when the numbers of event reporting node are less (20 nodes) is considerably high (Figure 5.11). As the number of event reporting nodes increases, sending congestion signals over multiple hops towards source nodes becomes difficult due to congestion and throughput decreases.

The throughput of proposed congestion control scheme without scheduling is better than source based congestion. At high number of event nodes due to interference, the throughput without scheduling is considerably low (Figure 5.11). Proposed congestion control scheme with scheduling is capable of maintaining its throughput. It efficiently handles congestion and packet drops due to inference.

The throughput observed from a 150 nodes sensor network with 50 event reporting nodes is shown in Figure 5.12.

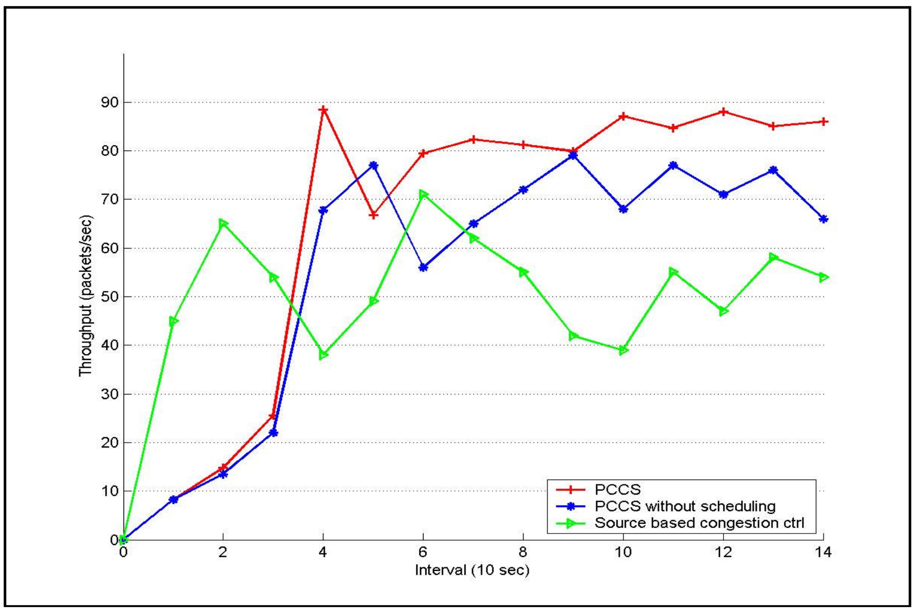

Figure 5.12 Throughput observed from 50 event reporting nodes using Proposed Congestion Control Scheme (PCCS) and source based congestion control.

In this simulation, source-based congestion control scheme uses an initial reporting rate of 10 packets per node per interval (10 sec). The proposed congestion control scheme uses an initial a lot length of 0.1 second which is assigned by first hop nodes to their previous hop nodes. Initially, the throughput of source-based congestion control is greater than proposed scheme (Figure 5.12).

A smaller value of slot length can give high throughput initially, but can result into congestion if number of event reporting nodes is higher. Since, the reporting rate of nodes is adjusted according to local network status instead of fixed increment/decrement factor (used by source based congestion control scheme), proposed congestion control scheme provides higher throughput.

Node arrangement can variably affect the overall throughput of the system. Shorter the hop distance of event reporting nodes from the destination or greater the number of first hop nodes, higher will be the throughput. In order to illustrate this fact, two different node arrangements are shown in Figure 5.13 and 5.14.

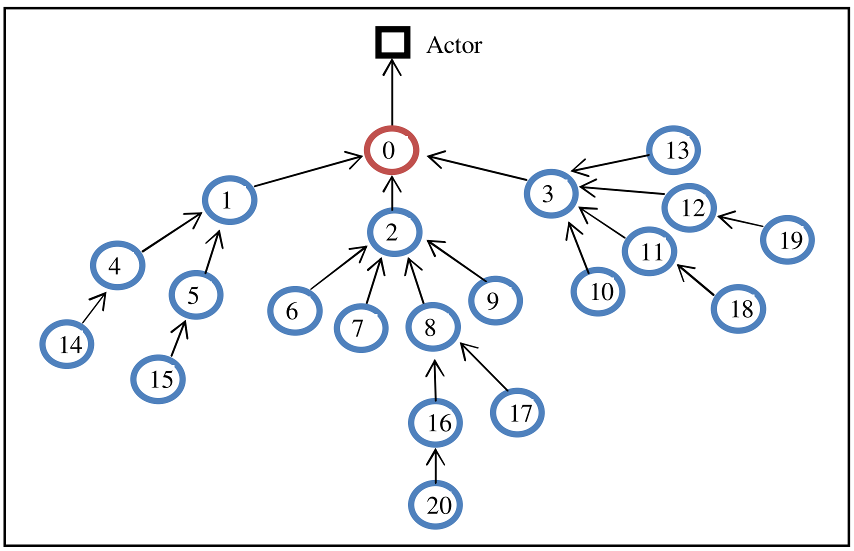

Figure 5.13 Arrangement of 20 event reporting nodes in a single flow (one first hop node).

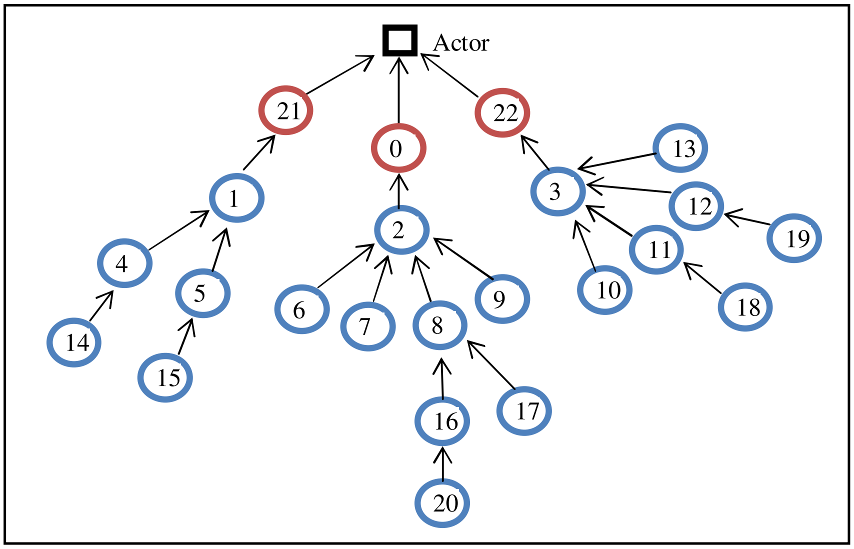

Figure 5.14 Arrangement of 20 event reporting nodes in three flows (three first hop nodes).

The number of event reporting nodes in both the Figures 5.13 and 5.14 is same. However, in Figure 5.13 single first hop node routes event information to destination while in Figure 5.14, three first hop nodes route event information to actor. For both these flows, the number of packets received at the actor during 140 second of event reporting is shown in Figure 5.15. By increasing the number of first hop nodes, the single flow shown in Figure 5.13 is divided into three flows, resulting the throughput of the system to be higher, in case of three flows.

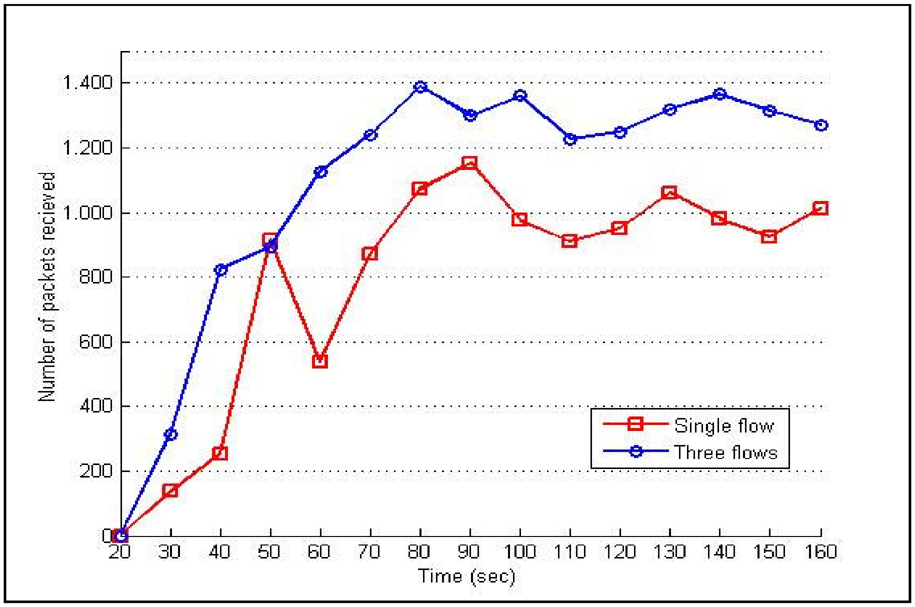

Figure 5.15 Number of packets received at the actor during 140 second of event reporting from 20 nodes arranged in a single and multiple (three) flows.

5.4.4 Energy Consumption

The residual energy of 130 sensor nodes with 50 event reporting nodes is shown in Figure 5.16. Initial energy of all the nodes is set to 0.1 Joules for this simulation. Proposed congestion control scheme handles congestion efficiently and also it does not increase the reporting rate of nodes more than they can handle. Therefore, despite of the additional scheduled packet transmission, proposed scheme decreases the energy consumption.

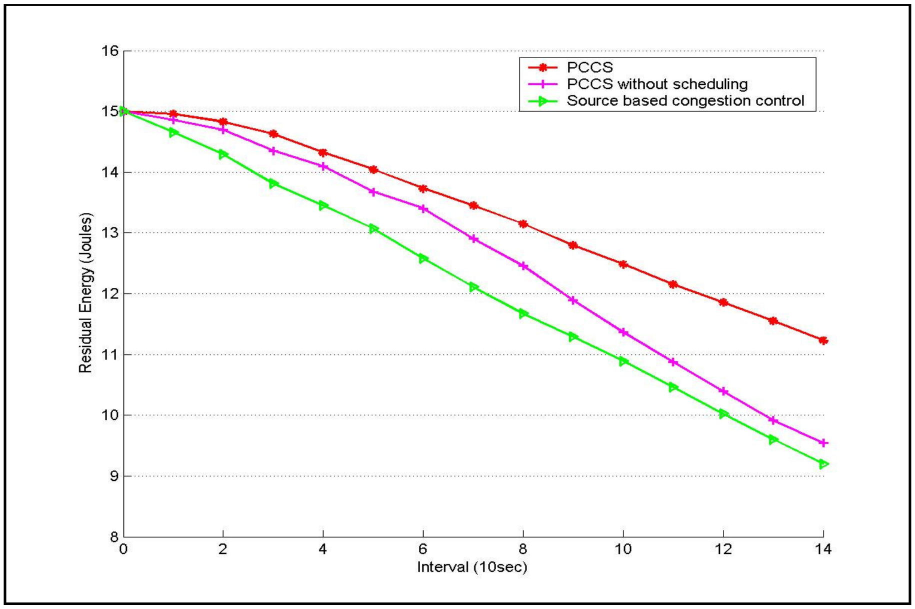

Figure 5.16 Residual energy of a 150 nodes network with Proposed Congestion Control Scheme (PCCS) and source based congestion control.

Proposed congestion control scheme, provides higher throughput and much less packet drops, its effectiveness is evident from Figure 5.17, in which the ratio of throughput observed over energy utilized is shown. Since in proposed congestion control scheme energy of nodes is efficiently utilized and throughput is high, the ratio is higher than other schemes in Figure 5.17.

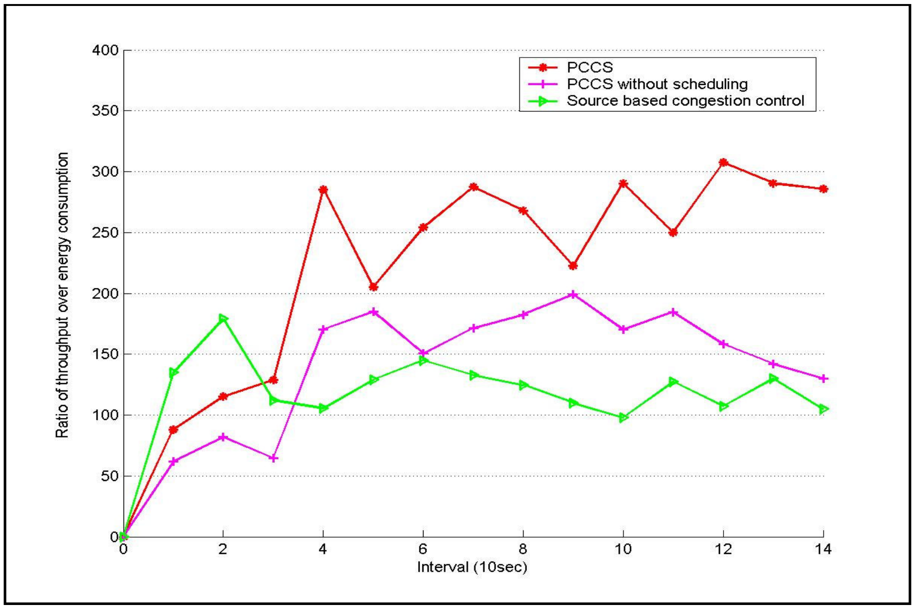

Figure 5.17 Ratio of throughput over energy consumed of 50 event nodes using Proposed Congestion Control Scheme (PCCS) and source based congestion control.

5.4.5 Length of Data Interval

The number of packets received using proposed congestion control scheme for different lengths of data interval is shown in Figure 5.18. The length of schedule interval is 1 second and event nodes are arranged as shown in Figure 5.13 in a 100x100m sensor field. All nodes are event reporting (*E-REP*) nodes while the first hop node is a simple routing node (*E-R*). Number of schedule packets generated during each schedule interval is 20.

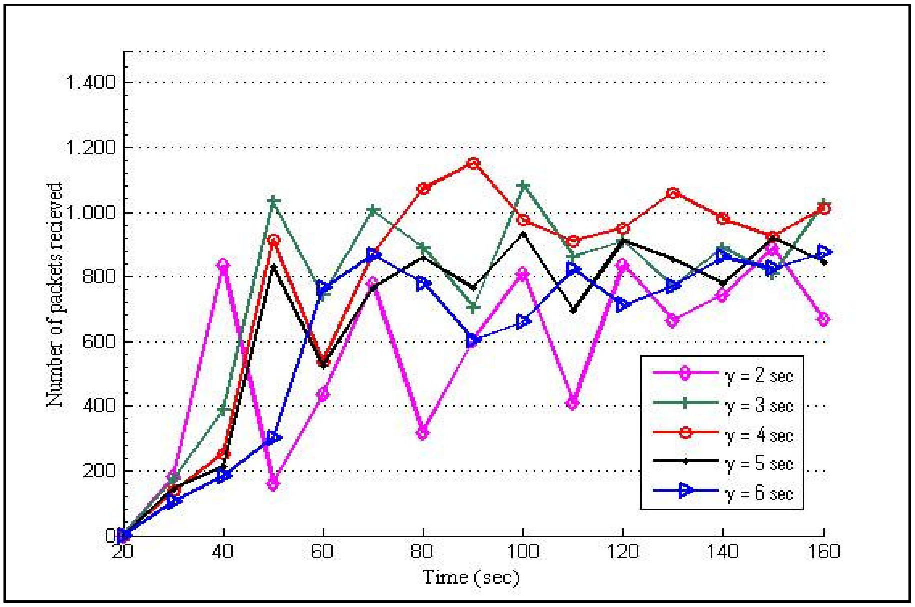

Figure 5.18 Number of packets received from 20 event nodes using proposed congestion control scheme with different data interval (γ) lengths.

It is evident from Figure 5.18 shorter the length of data interval, random the behavior of proposed congestion control scheme and longer the length of data interval, longer it will take to achieve optimal throughput. The overhead of schedule packets is calculated in terms of total schedule packet generated and total data packets delivered to actor; assuming the length of both packets to be the same. The overheads of maintaining schedule packets while using 3, 4, 5, and 6 seconds of data interval are shown in Table 5.3.

Table 5.3 Percentage of overhead calculated during 140 seconds of event reporting using proposed congestion control scheme with different slot lengths.

Data interval length (sec)	**Total packets received at destination**	**Total number of schedule packet transmissions**	**Percentage of overhead**
2	8346	920	11.02
3	11283	700	6.20
4	11750	560	4.76
5	10040	460	4.58
6	9124	400	4.38

Increasing the length of data interval above 6 seconds decreases the overhead of proposed scheme, but results in lower throughput with higher packet drops.

Decreasing slot length below 3 seconds results into more random behavior and increases schedule packet overhead. Therefore, the proposed scheme uses data interval length of 4 seconds in order to provide maximum throughput with low overhead.

The number of schedule packets generated and data packets delivered to the actor during consecutive data and schedule intervals is shown in Figure 5.19.

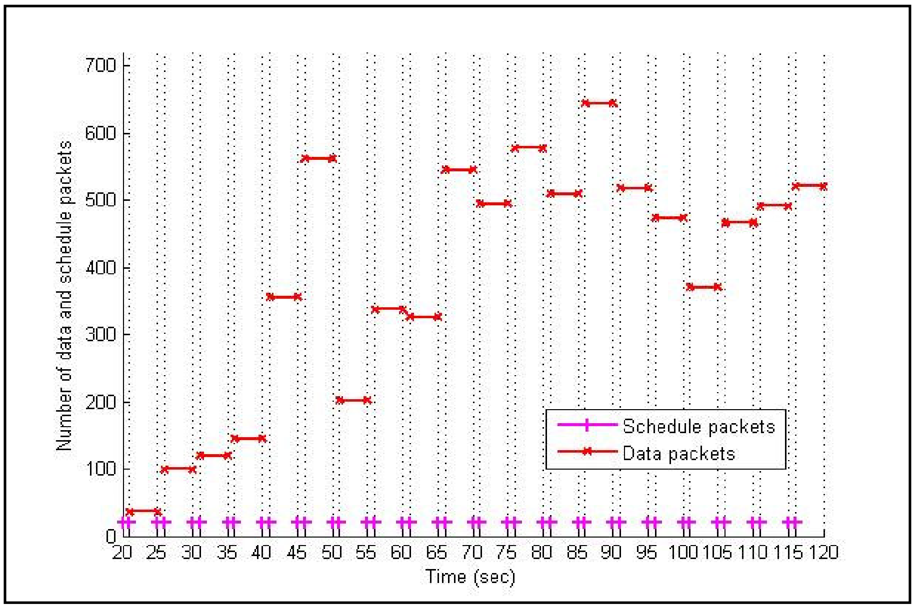

Figure 5.19 Number of schedule packets generated and data packets delivered to the actor using proposed congestion control scheme.

The length of data interval is 4 seconds while schedule interval length is 1 second (Figure 5.19). The simulation uses the node arrangement as shown in Figure 5.13. Since the proposed scheme uses average hop-by-hop packet delivery time to adjust the reporting rate of nodes, per data interval throughput varies in Figure 5.19. Therefore, the proposed scheme in each schedule interval adapts a reporting rate that optimally utilizes the buffer of nodes (to provide maximum throughput) while avoiding congestion.

5.4.6 Multiple Actors

The overall throughput observed at a single and multiple (two) actors receiving event information from 20 event reporting nodes is shown in Figure 5.20. The total number of non event reporting nodes in the network is 100. The event reporting nodes are randomly deployed in an event region centered at (50, 50) coordinates with a event region radius of 20 meters.

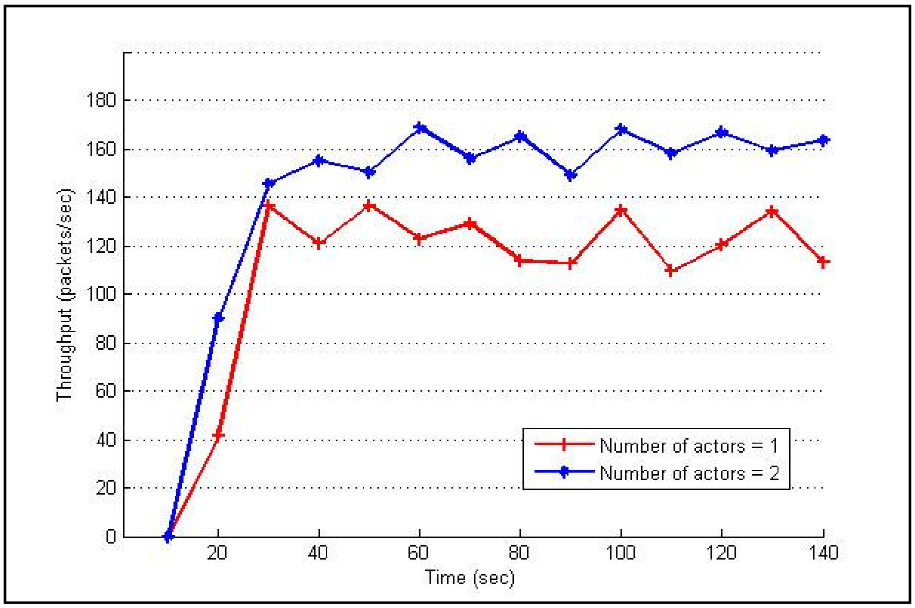

Figure 5.20 Total throughput observed from a network with either one or two actors using proposed congestion control scheme.

The event information is reported by more than one first hop nodes to the actors. Initially, the throughput is observed with a single actor placed at (80, 50) coordinates using proposed congestion control scheme. For two actors case another actor is introduced at (20, 50) coordinates. From figure 5.20, it is evident that the throughput increases in case of multiple actors because of the increase in the number of event flows and also due to the decrease in the hop distance of nodes from the destination.

CHAPTER SIX
SIMPLE, FAIR AND PRIORITIZED RELIABLE EVENT TRANSPORT MODES

6.1 Motivation

Wireless sensor networks are characterized by their unique requirements which are application specific. An application may require general event region information, per node event information, or multiple events information etc. In the mining application of wireless sensor and actor network (WSAN) discussed in section 3.2.1, for events like increase in temperature and pressure, general event region information can be sufficient but for events like leakage of poisonous gas or oxygen content in air, precise per node information is required to identify unsafe regions in the mine. Inter-related events like fire and decrease in oxygen content in air can occur at the same time. Multiple events may need to be reported at different rates when observed together. Therefore, a single application might require information to be delivered in different event reporting modes e.g., simple, fair or prioritized.

The proposed sensors-to-actors transport is divided into different event transport modes as mentioned in section 4.2. These transport modes use the proposed congestion control scheme for mitigating congestion and schedule based rate adjustment scheme presented in previous chapter. In existing literature, these modes are implemented for different applications and different basic assumptions (Akan & Akyildiz, 2005; Tien & Bajcsy, 2004; Wang, & et al., 2007). However, the proposed transport modes are implemented in modular fashion so that they can be used either independently or in a combined fashion.

6.2 System Model

In order to define the system model in detail, the basic system related definitions and transport header structure for the proposed simple, fair and prioritized transport modes should be explained. The basic system model including node types and next hop table setup is similar to proposed congestion control scheme presented in section 5.2.

The sensors-to-actors transport is triggered by events. Sensor nodes after detecting an event send information to the actor via intermediate nodes. Initially event nodes transmit an *event packet* which contains event identification number, total reporting rate, sub-tree size of the node and the mode of transport for the event. If an event reporting node (*E-REP*) receives an event packet, then it changes its status to event reporting & routing (E-REP-R) node. Likewise, if an intermediate idle node after receiving an event packet changes its status to routing node (E-R).

After receiving an event packet, nodes update their total reporting rates and transmit the new total reporting rate to their next hop node in the *event packet* until it reaches the destination (actor). Also, the intermediate nodes maintain a previous hop table in order to determine the reporting rate and sub-tree size of previous hop nodes.

For the sake of simplicity, it is assumed that all nodes shown in Figure 6.1 are event reporting nodes, except node *a*. Furthermore, let the initial reporting rate of nodes be *b(10), c(10), d(10), e(10), f(20), g(10)*. Then, the total reporting rate of node *b, c* and *d* will be 10, 50 and 10 respectively, while node *a* has a total reporting rate of 70.

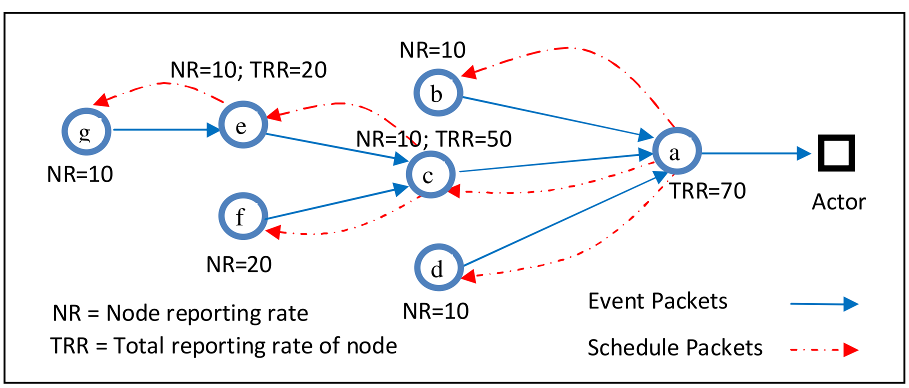

Figure 6.1 The flow of event and schedule packets with intial Node reporting Rate (NR) for each event reporting node and Total Reporting Rate (TRR).

The sub-tree size of nodes *b, c* and *d* are 1, 4 and 1 respectively, while node *a* has total sub-tree size of 6 (Figure 6.1). The previous hop tables maintained by nodes *a* and *c* are shown in Tables 6.1 and 6.2.

Table 6.1 Previous hop table maintained by node *a*.

Node ID	Reporting	Subtree size
B	10	2
C	50	4
D	20	1

Table 6.2 Previous hop table maintained by node *c*.

Node ID	Reporting	Subtree size
e	20	2
f	20	1

The transport mode in which nodes report the event can be either predefined in the event definition or can be set by the actor node. If transport mode is predefined, then nodes according to event identification number can select appropriate transport mode. In case of actor based selection, event nodes first send *event packets* to the actor, which then decides the mode of transport. For real-time events, considerable amount of time is lost to first send event information to actor and then to update all event nodes about the transport mode. As a result, it is assumed in this study that all nodes are aware of the transport mode and this information is available in event definitions.

The transport headers for different packets used by these transport modes are shown in Tables 6.3, 6.4 and 6.5.

Table 6.3 Transport header for event packet.

2 Bytes	2 Bytes	2 Bytes	2 bits	2 Bytes	2 Bytes	2 Bytes	2 Bytes
Source ID	Destination ID	Actor ID	Packet type	Event ID	Reporting rate	Sub-tree size	Header length

Table 6.4 Transport header for data packet.

2 Bytes	2 Bytes	2 Bytes	2 bits	2 Bytes	4 Bytes	2 Bytes
Source ID	Destination ID	Actor ID	Packet type	Event ID	Time stamp	Header length

Table 6.5 Transport header for schedule packet.

2 Byte	2 Bytes	2 bits	2 bits	4 Bytes	2 Bytes	2 Bytes	2 Bytes	2 Bytes
Source ID	Destination ID	Mode type	Packet type	Slot length	Start slot	End slot	Total slots	Header length

The explanation of the fields used in the data, schedule and event packet headers is as follows:

- *Source ID:* The field contains identification number of node sending the packet since in sensor networks nodes are generally referenced with unique identification numbers (like 0,1,2,3 and so on) instead of IP addresses.
- *Destination ID:* For data and event packets this field contains next hop nodes identification number while for schedule packets it contains child node identification number.
- *Actor ID:* The identification number of closest actor from the source node.
- *Packet type:* It identifies the packet to be one of event, data and schedule packets.
- *Header length:* The length of proposed transport header.
- *Time stamp:* This field contains a time stamp at which the data packet is handed to the routing layer. The field is used to calculate hop-by-hop packet delivery time for data packets.
- *Event ID:* Each event has a unique identification number and the field contains the Identification number of the event.
- *Reporting rate:* The field contains the total reporting rate of a node and is used in event packets only. In case of E-REP nodes it is equal to the initial reporting rate of the event being reported by these nodes. For E-R nodes is equal to the sum of initial reporting rate of the nodes routing through these nodes while in case of E-REP-R nodes it equal to the sum of the both node's initial reporting rate and the rate of nodes routing through these nodes.
- *Sub-tree size:* The field contains the sub-tree size of the node sending the event packet.
- *Mode type:* Nodes select mode of transport depending on the event Id, however, in case of actor based transport mode selection this field contains

the selected mode of transport (normal, fair, prioritized, real time) by the actor.

- *Slot length:* The field specifies the duration of a slot length, allocated by a nodes to their previous hop nodes.
- *Start:* Start slot number of allocated slots for the next data interval.
- *End:* End slot number of allocated slots for the next data interval.
- *Total:* Total number of slots allocated by next hop node to all previous hop nodes for the next data interval.

Simple Event Transport Mode (SETM) aims to reliably transport general event region information to the destination. Event based wireless sensor and actor networks (WSANs) are required to reliably detect an event from the sensor field and most of the applications require general event information irrespective of per event node contribution at the destination. Simple event transport can be achieved by avoiding congestion and adjusting reporting rate of nodes in order to achieve application based reliability or maximum throughput. Therefore, SETM uses the proposed congestion control scheme for congestion avoidance and rate control.

Sensors-to-actors information flow is characterized by number of individual flows. The overall number of packet received at the destination during an interval by all these flows is the throughput observed at the destination (during the interval). A number of actors can act as destination at the same time. Hence, the sum of throughput observed at all the actors is the systems throughput. If the number of packets received in the interval by all the actors are not equal to the required (application specified) number of packets, then in order to achieve application based reliability, the actors informs their first hop nodes to adjust reporting rate (slot length).

Fair Event Transport Mode (FETM) aims to provide same per node throughput at the destination from a single flow. Fairness can be achieved by avoiding congestion and by assigning same reporting rate to all event reporting nodes. Also, a packet forwarding scheme is required to ensure fair event packet delivery from a number of sources which can be at multiple hop distance from a single destination.

As a summary, important issues concerning fairness are congestion control, fair rate adjustment and packet delivery. In this study, the FETM uses proposed congestion control scheme for congestion mitigation, slot length allocation is done on the basis of sub-tree size for fairly distributing reporting rate among event node and the proposed schedule based packet forwarding scheme has been modified to provide fairness.

Prioritized Event Transport Mode (PETM) is designed to handle multiple events according to their application defined reporting rates. It distributes the flow bandwidth among different event reporting nodes according to their initial reporting rates. Thus, the nodes with high reporting rates obtain more priority and deliver more packets to the actor than nodes with lower reporting rates.

PETM in other words is an event based fairness transport mode, as compared to FETM which is a node based fairness transport. In node based fairness, the destination receives same number of packets from all the nodes reporting either same or different events, because the rate allocation is done on the basis of sub-tree size; irrespective of the reporting rate of nodes. As a result, node based fairness is unable to handle multiple events with different reporting rate requirements.

On the other hand, in PETM (or event based fairness) the actor receives same number of event packets from all the nodes reporting the same event. If all event nodes are reporting the same event, then all event nodes will get equal share of the link bandwidth resulting into fairness; similar to node based fairness (or FETM). In case of multiple events, nodes according to initial event reporting rate will get share of the link bandwidth.

6.3 Operation of Simple, Fair and Prioritized Transport Modes

The transport modes operate on top of proposed congestion control and schedule based scheme to provide reliable transport. The transport mode use the same slot calculation procedure presented in section 5.3.1, for detecting and removing congestion. Also, the operation of schedule based scheme is similar to the one presented in chapter 5. However, depending on the selected mode, the slot allocation procedure differs. Slots allocation and the procedure for achieving application defined reliability will be explained.

A slot is a time interval during which a node can forward a single packet. Greater the number of slots assigned to a particular node greater will be its reporting rate. In case of simple and prioritized event transport modes, slots are assigned equal to the total reporting rate of a node. In fair event transport mode, slots are assigned by nodes, which are equal to their sub-tree size.

- SETM assigns slots to nodes with respect to total reporting rate (TRR) traversing through a node and the minimum reporting rate observed at the nodes. The slot allocation is similar to the one explained in section 5.3.2.
- FETM assigns slots to nodes according to their sub-tree size. The sub-tree size depends on number of event nodes not on their reporting rates. Even in case of multiple events with different reporting rates, nodes can forward packets with node based fairness so that all nodes have same representation at the destination. The number of slots S_i^j assigned to the j^{th} previous hop neighbor of node i is equal to the sub-tree size T_j of the j^{th} node.
- PETM assigns slots to nodes with respect to TRR and the minimum reporting rate among all the events observed by the network (R_{min}). In case of prioritized event transport, total number of slots allocated to the previous hop nodes (N_k) of i will be $\sum_{j=1}^{k} R_j/R_{min}$.

Depending on the system given in Figure 6.1, the slot allocation by node *a* in SETM and PETM modes is shown in Table 6.6 and slot allocation by node *a* in FETM is shown in table 6.7.

Table 6.6 Transmission schedule generated by node *a* in simple and prioritized event transport.

Node ID	Total Slots	Initial Slot	End Slot	Slot length (seconds)
B	7	1	1	0.1
C	7	2	6	0.1
D	7	7	7	0.1

Table 6.7 Transmission schedule generated by node *a* in fair event transport mode.

Node ID	Total	Initial Slot	End Slot	Slot length (seconds)
B	6	1	1	0.1
C	6	2	5	0.1
D	6	6	6	0.1

Nodes *b, c* and *d* will divide their data interval into 0.1 second intervals; the initial slot length on event occurrence. These nodes will forward one event packet to node *a* during their allocated slots. Since, nodes *c* is an E-REP-R node therefore depending on event reporting mode node *c* will generate a schedule for nodes *e* and *f*. The schedules given to nodes *e* and *f* by node *c* in SETM, FETM and PETM respectively are given in Tables 6.8, 6.9 and 6.10. Likewise, node *e* will generate schedules for child node *g*, which are shown in Tables 6.11, 6.12 and 6.13 for SET, FETM and PETM respectively.

Table 6.8 Transmission schedule generated by node *c* in simple event transport mode.

Node ID	Total	Initial Slot	End Slot	Slot length (seconds)
e	4	1	2	0.1
f	4	3	4	0.1

Table 6.9 Transmission schedule generated by node *c* in fair event transport mode.

Node ID	Total	Initial Slot	End Slot	Slot length (seconds)
e	6	3	4	0.1
f	6	5	5	0.1

Table 6.10 Transmission schedule generated by node *c* in prioritized event transport mode.

Node ID	Total Slots	Initial Slot	End Slot	Slot length (seconds)
e	7	3	4	0.1
f	7	5	6	0.1

Table 6.11 Transmission schedule generated by node *e* in simple event transport mode.

Node ID	Total Slots	Initial Slot	End Slot	Slot length (seconds)
g	1	1	1	0.1

Table 6.12 Transmission schedule generated by node *e* in fair event transport mode.

Node ID	Total Slots	Initial Slot	End Slot	Slot length (seconds)
g	6	4	4	0.1

Table 6.13 Transmission schedule generated by node *e* in prioritized event transport mode.

Node ID	Total Slots	Initial Slot	End Slot	Slot length (seconds)
g	7	4	4	0.1

The reliability level is observed by the coordinated effort of actors after every interval is *network observed reliability* (OR_{NET}). Actor(s) in the proposed sensors-to-actors transport achieves *required reliability* (RR) by sending different messages to the first hop nodes. At the end of each reliability interval, the reliability can be low, high or equal to the required reliability. These three possibilities will be discussed below:

1. **OR_{NET} < RR:** If the observed reliability is less than required reliability, actor(s) broadcasts increase rate message. Member nodes of the actor automatically adjust their reporting rates to achieve maximum throughput. Since, the nodes adjust reporting rate according to network conditions, the

actors do not send an increment factor to the first hop nodes. The amount of increment is dependent on network conditions. If the network is not congested and node density is low, then the hop-by-hop packet delivery time will be less resulting into smaller slot length for the next interval or in other term greater throughput. In case of congestion, slot length increases and reporting rate decreases. The proposed sensors-to-actors transport does not provide any congestion information to the actor. Since, the actor is unaware of the network situation, it is inappropriate for the actor to send increment factor.

There are three basic reasons for decoupling the congestion control mechanism from the actor. First, actor based congestion control decreases the reporting rate of whole network even if a single node is congested. But in sensor networks, where nodes can have different buffer sizes, decreasing the reporting rate of all the nodes decreases the throughput of the network considerably. Second, actor based congestion control is slow to respond to a congestion. Since an actor broadcasts new reporting rate at the end of each reliability interval, whereas, congestion can occur at any time during the interval. Third, in case of multiple events with different reporting rates, congestion occurs due to an event requiring high reporting rate. However, actor based congestion control decreases the reporting rate of all the nodes; resulting into a decrease in the reliability of all the events.

2. **OR_{NET} > RR:** If the network observed reliability is greater than required reliability, actor(s) send decrease reporting rate message (along with a percentage of decrease) to the first hop nodes at the end of interval. The actor calculates the percentage of decrease required to achieve application defined reliability. This percentage is divided among the first hop nodes according to their share in the throughput at the actor. The first hop nodes decrease their reporting rates by increasing their slot lengths according to the actor specified percentage of increase. Hence nodes decrease their reporting rate until the required reliability level is achieved. This ensures that the nodes do not over exert themselves by reporting at a rate greater than the required rate.

3. **OR$_{NET}$ ≈ RR:** In this case, network observed reliability is approximately equal to required reliability. Therefore, the actor(s) broadcasts maintain reliability message to the first hop nodes which maintain their slot lengths in order to achieve same throughput in the next interval.

The following procedure is called by first hop nodes, on the reception of actor message for the selection of an appropriate slot length, in order to achieve application defined reliability.

Let λ_i^t be the slot length sent by a node i to its previous hop nodes for the t^{th} interval. Moreover, at the end of t^{th} interval let λ_i be the calculated slot length by the i^{th} node while λ_i^{t+1} is the slot length sent by the i^{th} node for the $(t+1)^{th}$ interval.

```
1. // Maintain reporting rate message received
      If ACTOR_MESSAGE = MAINTAIN
         NEXT_SLOT_LENGTH (λ_i^{t+1}) = PREVIOUS_SLOT_LENGTH (λ_i^t)
         GO TO STEP 4

2. // Decrease reporting rate message received
      If ACTOR_MESSAGE = DECREASE
      //DEC_PERCENTAGE; percentage of decrease sent by actor node
          ADJUST_FACTOR = (λ_i^t x DEC_PERCENTAGE) / 100
          λ_i^{t+1} = ADJUST_FACTOR + λ_i^t
          If λ_i^{t+1} < CALCULATED_SLOT_LENGTH (λ_i)
             λ_i^{t+1} = λ_i

             GO TO STEP 4
          Else
             GO TO STEP 4

3. // If increase message received from an actor send
   calculated slot length
      If ACTOR_MESSAGE = INCREASE
         λ_i^{t+1} = λ_i
         GO TO STEP 4

4. TRANSMIT SCHEDULE
```

6.4 Simulation Results

Proposed transport modes are evaluated in terms of achieving application defined reliability, packet delivery ratio, throughput and energy consumption under different nodes arrangements. The simulation results for simple, fair and prioritized event transport modes will be discussed in detail.

6.4.1 Simple Event Transport Mode (SETM)

SETM is similar to proposed congestion control scheme in terms of operation hence the simulation results are similar to the ones discussed in section 5.4. Like other transport modes SETM provides reliable information transport. All transport modes achieve reliability in similar fashion therefore it is only mentioned in this section.

Simulation results for application defined throughput of 300, 500, 700 and 1400 packets per 10 second interval observed at the actor are shown in Figures 6.2 and 6.3. The node arrangement shown in Figure 5.13 is used in these simulations.

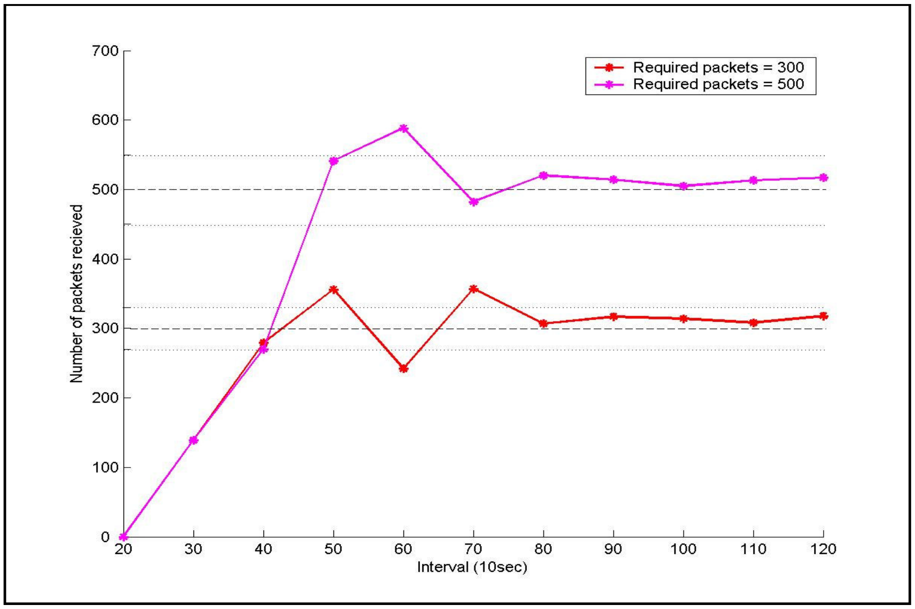

Figure 6.2 Achieving application defined throughput of 300 and 500 packets per actor interval from 20 event nodes using Simple Event Transport Mode (SETM).

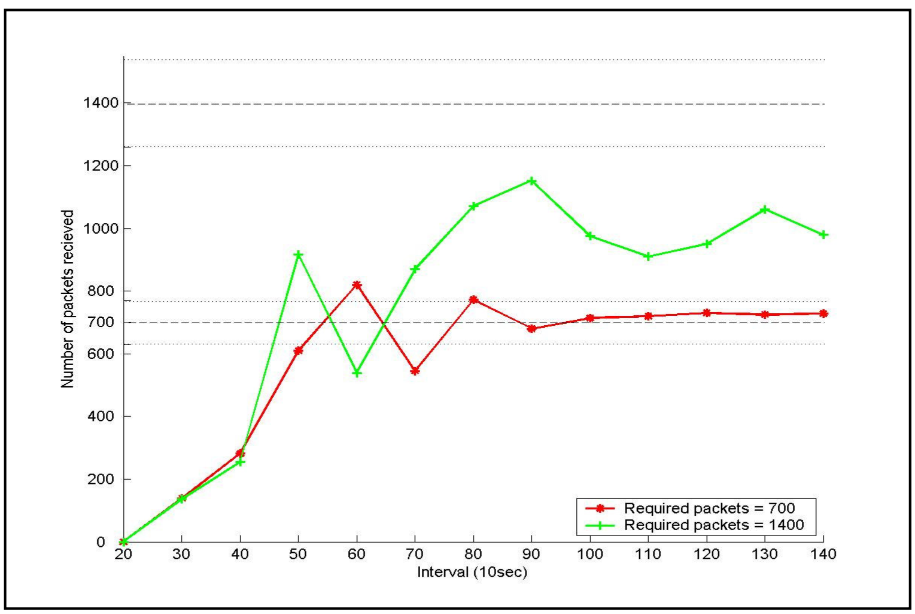

Figure 6.3 Achieving application defined throughput of 700 and 1400 packets per actor interval from 20 event nodes using Simple Event Transport Mode (SETM).

If application defined throughput (300, 500 and 700 packets per actor interval) is less than maximum achievable throughput (approximately 1200 packets per actor interval for the node arrangement shown in Figure 5.13), then the proposed transport modes can achieve application's defined reliability. However, if the maximum achievable throughput is less than application's defined throughput (e.g., 1400 packets per actor interval), then even if the actor broadcast increase reporting rate messages, nodes can not further increase their reporting rates due to congestion.

In this case, SETM and other transport modes try to achieve maximum throughput and continuously adjust their reporting rates which results into a wavy behavior as shown in Figure 6.3, for required throughput of 1400 packets. In order to further explain this fact, the simulation for required throughput of 1400 packets (shown in Figure 6.3) is modified, by adding extra nodes in the event region.

The nodes initially report event according to the node arrangement shown in Figure 5.13. After 120 seconds of event reporting, 15 more nodes join the event nodes, such that the maximum achievable throughput increases above required throughput (1400 packets per actor interval). The simulation result is shown in Figure 6.4, where for first 120 second of event reporting nodes provide maximum throughput.

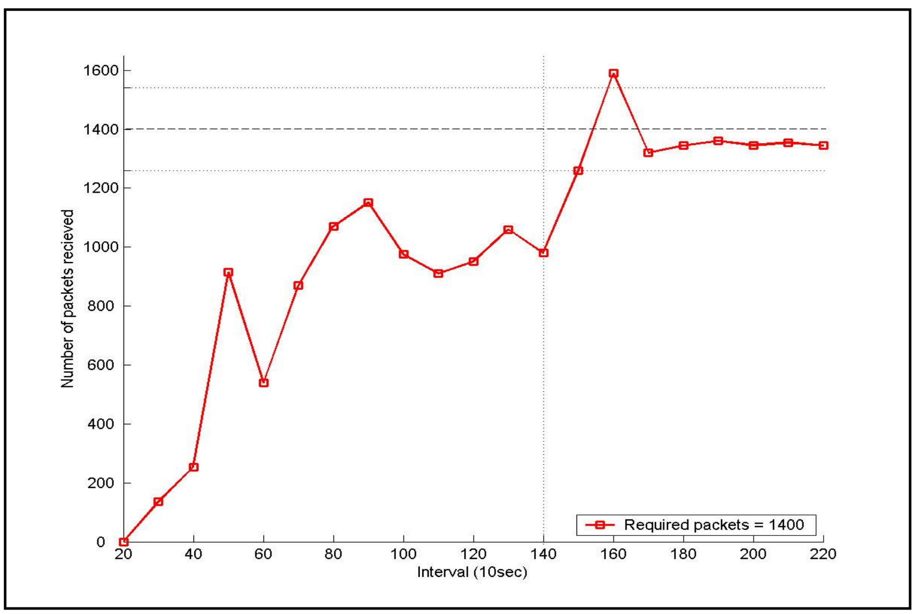

Figure 6.4 Achieving application defined throughput of 1400 packets after adding 15 nodes (at 140sec) to 20 event nodes using simple event transport mode.

Once extra nodes join the event reporting nodes, topology changes and the network becomes capable of providing more throughputs. Hence, SETM achieves the required reliability as soon as maximum achievable throughput increases above required throughput (Figure 6.4).

In WSANs nodes can stop reporting event due to battery failures, physical damages, upon destination's instruction and on the removal of event from a particular region. The behavior of SETM in such conditions, when application defined throughput is achieved and some nodes stop reporting event is shown in Figure 6.5. The situation is similar to low reliability case as the throughput decreases to below of the required level.

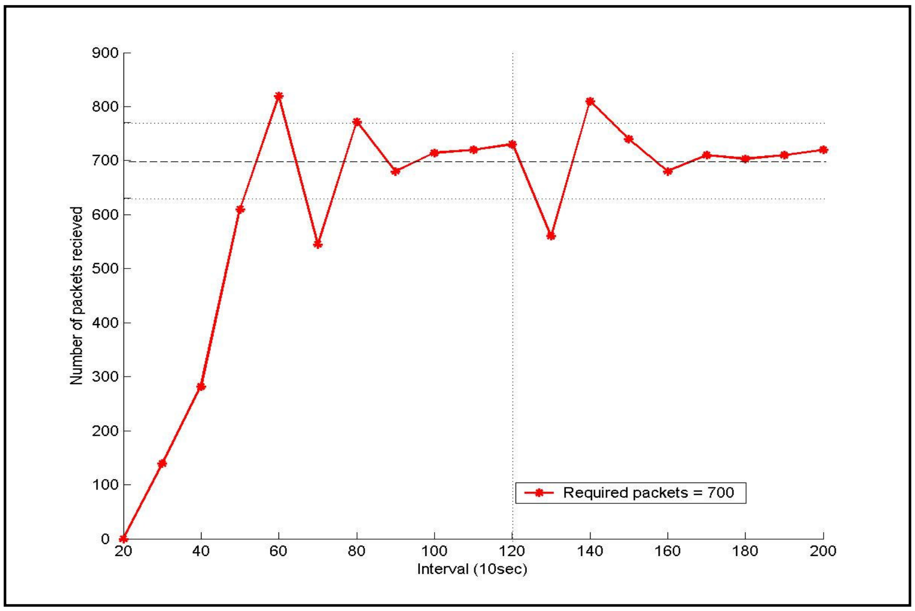

Figure 6.5 Re-achieving application defined throughput of 700 packets after deleting 10 nodes from 20 event nodes (at 120sec) using simple event transport mode.

Nodes are arranged as shown in Figure 5.13 and the required number of packets per actor interval is 700 (Figure 6.5). After 100 seconds of event reporting, ten nodes (11-20) stop reporting event. As a result, the throughput observed decreases and the actor stops broadcasting maintain reporting rate message. Event nodes increase their reporting rates, as they receive increase message from the actor, until required reliability is achieved.

6.4.2 Fair Event Transport Mode (FETM)

If the per node throughput at the actor is same for all event reporting nodes in the flow, then event reporting is fair. The simulation scenario where 20 nodes are reporting the same event through first hop node 0 to the actor is shown in Figure 6.6.

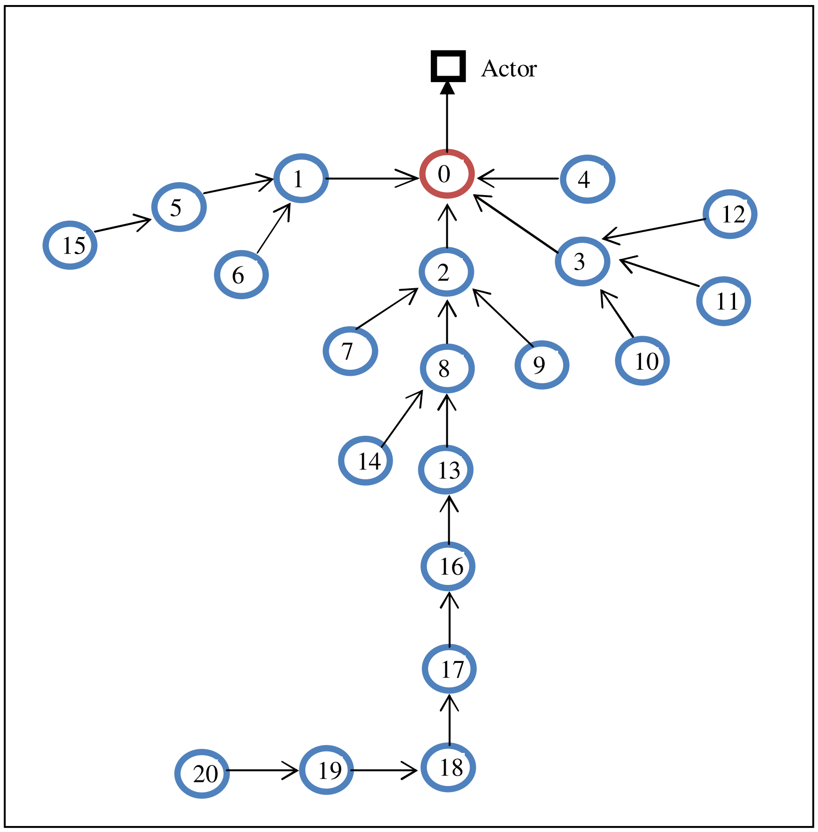

Figure 6.6 Arrangement of 20 event reporting nodes in a single flow with 9 hops in a 100x100m sensor field with only single hop interference.

In order to decrease multiple hop interference, the radio range used in the simulation scenario is 10meters (Figure 6.6). The arrow heads show the direction of minimum hop routing. The actor is located at coordinates (90,50) in a 100x100 sensor field with maximum hop distance of 9 hops.

In this study, FETM is compared with Congestion Control with Fairness (CCF) (Tien & Bajcsy, 2004) and Simple Event Reporting scheme (SER). CCF is a commonly referenced fairness scheme for wireless sensor networks. In CCF, nodes use packet service time to predict their reporting rates. Furthermore, nodes use buffer size to predict congestion. Each node implements a separate child node queue at the transport layer. Fairness is achieved by forwarding packets from the child node queues equal to sub-tree size of each child node.

Simple Event Reporting (SER) scheme, uses source based congestion control and an Additive Increase Multiple Decrease (AIMD) rate adjustment policy. It uses buffer occupancy for congestion control with initial reporting rate of 2 packets per second by event nodes. Reporting rate is increased by a factor of 1.2 after every actor interval (10 sec) and decreased to half in case of congestion by source nodes.

This study implements simple event reporting to check the affect of per node throughput at the destination from source nodes which are at multiple hop distance from the actor; without any fairness scheme.

As it can be seen from Figure 6.7, simple event reporting results into variable per node throughput because it does not use any explicit fairness mechanism. CCF (Tien & Bajcsy, 2004) provides considerably fair output but FETM provides even better results with high per node throughput, because in FETM, each node is assigned a schedule in order to assure fair per node throughput.

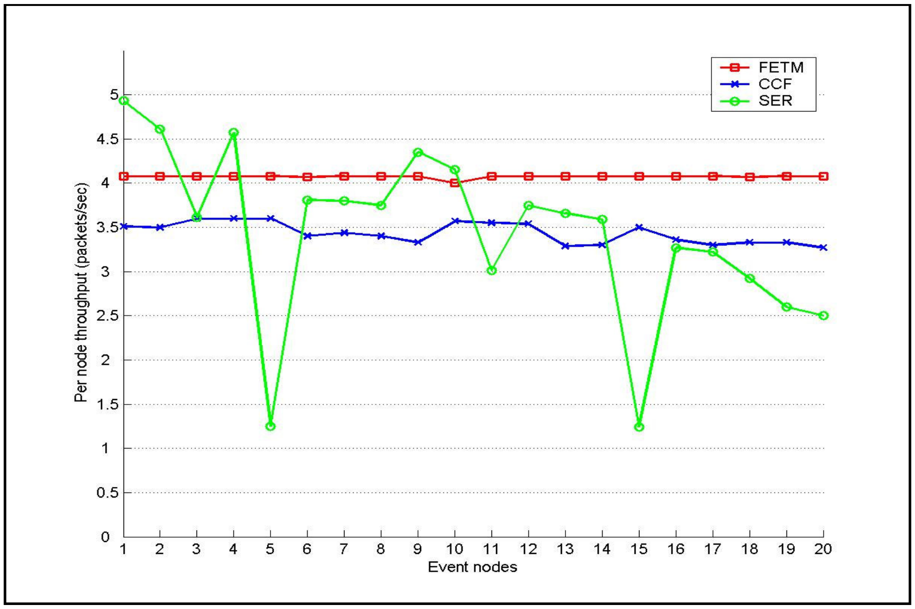

Figure 6.7 Per node throughput of 20 event reporting nodes using Fair Event Transport Mode (FETM), Congestion Control with Fairness (CCF) and Simple Event Reporting (SER).

To further compare the per node throughput of FETM and CCF the following single hop interference and multiple hop interference scenarios are used.

1) *Single Hop Interference:* For evaluating the performance of FETM in single hop interference, event nodes are placed in an event region centered at coordinates (60,60). The diameter of event region is 10 meters. All nodes are arranged so that they are at a single hop distance from the first hop node. The actor is at coordinates (90, 90). The per node throughput of 50 and 100 event reporting nodes using FETM and CCF are shown in Figures 6.8 and 6.9. FETM provides high per node throughput than CCF in both node densities. Since packet drops due to interference increases, the performance of CCF severely degrades in high density (100 event nodes).

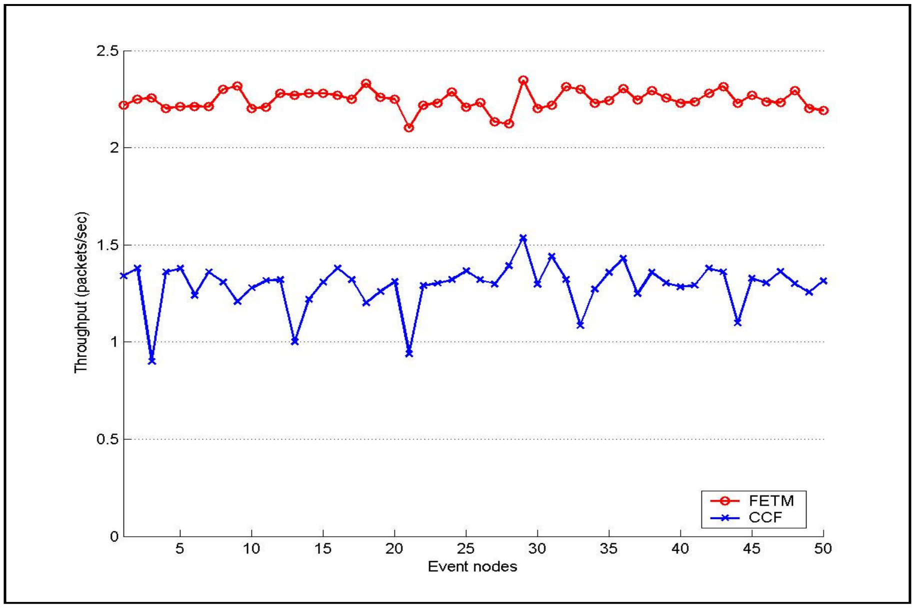

Figure 6.8 Per node throughput of 50 event nodes arranged on same hop and reporting with Fair Event Transport Mode (FETM) and Congestion Control with Fairness (CCF).

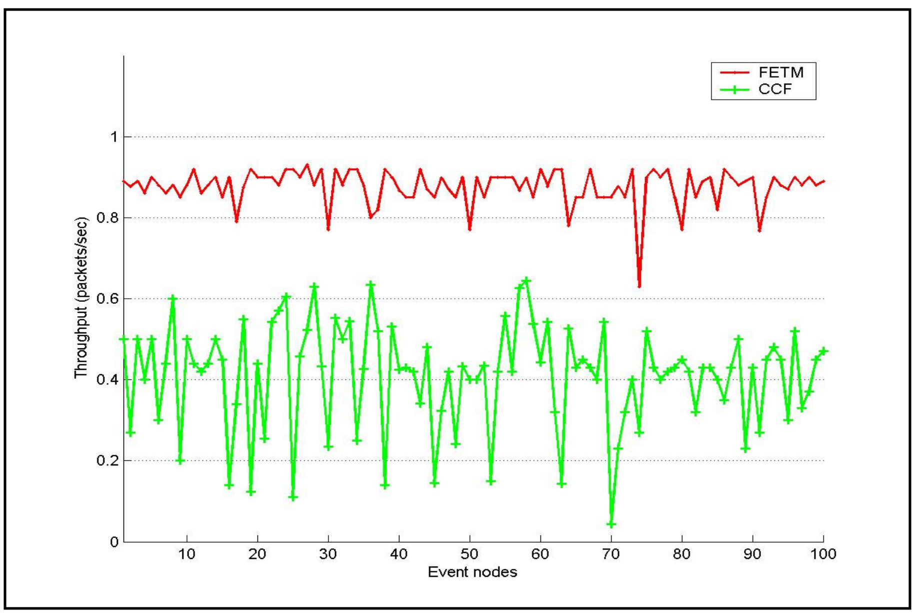

Figure 6.9 Per node throughput of 100 nodes arranged on same hop and reporting event with Fair Event Transport Mode (FETM) and Congestion Control with Fairness (CCF).

2) *Multiple Hops Interference:* In this case, the event nodes are randomly placed in an event region centered at coordinates (40,40). The event region has a diameter of 40 meters and the actor is at coordinates (90, 90). 50 and 100 event nodes report event to the actor through a single first hop node. It is evident from Figures 6.10 and 6.11 that FETM in both node densities, provide more fair and high per node throughput than CCF.

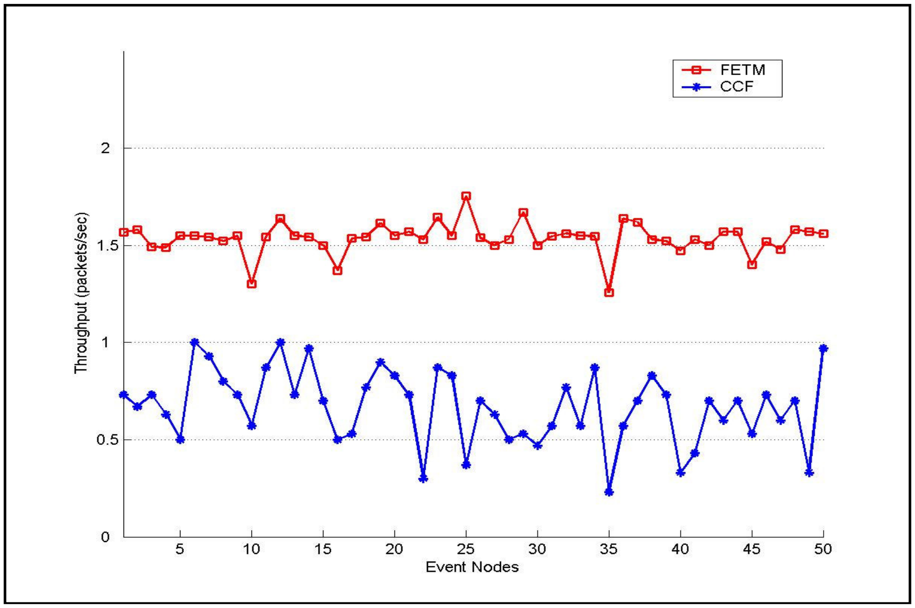

Figure 6.10 Per node throughput of 50 event nodes randomly deployed and reporting with Fair Event Transport Mode (FETM) and Congestion Control with Fairness (CCF).

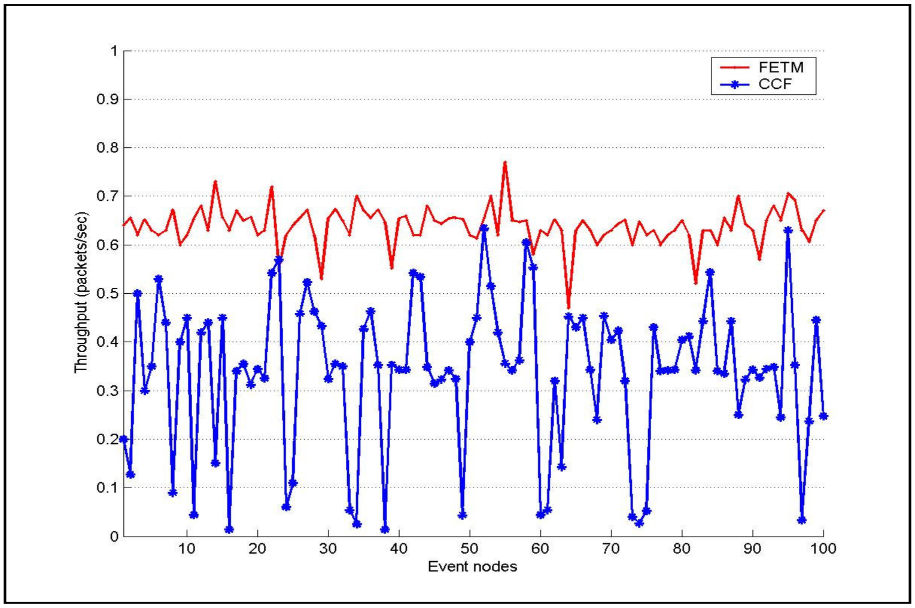

Figure 6.11 Per node throughput of 100 event nodes randomly deployed and reporting with Fair Event Transport Mode (FETM) and Congestion Control with Fairness (CCF).

The average per node throughput of FETM with SETM, CCF and SER, at the actor is shown in Figure 6.12. Multiple hops interference simulation scenario is used and after 70 seconds of event reporting 20 more nodes starts reporting the event from the same event region.

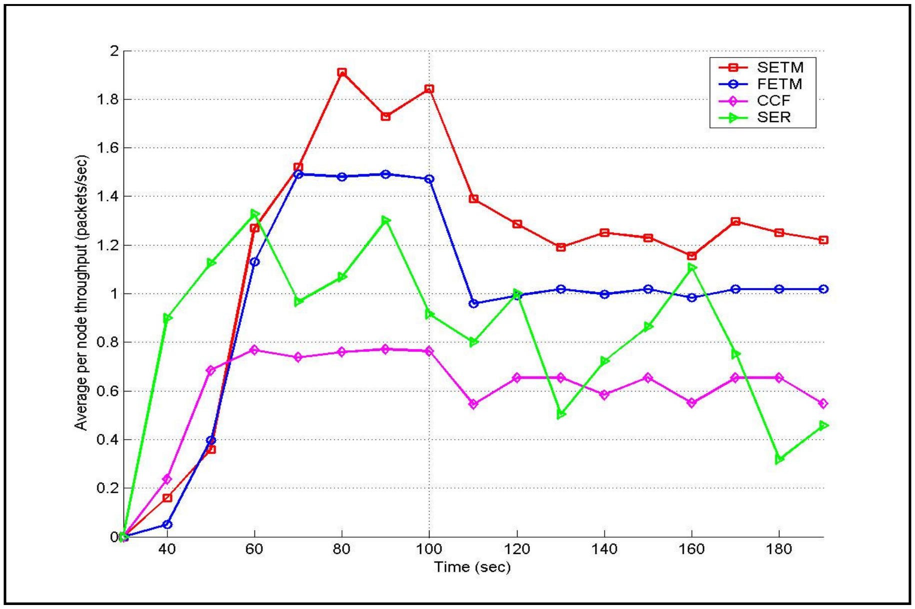

Figure 6.12 Average per node throughput observed during 70 second of event reporting from 50 event nodes and later from 20 extra event nodes for the same event region; using simple event transport mode (SETM), fair event transport mode (FETM), congestion control with fairness (CCF) and simple event reporting (SER).

Since the AMID rate control scheme operates irrespective of number of event reporting nodes, event reporting with SER scheme shows variable average per node throughput (Figure 6.12). Also, sending congestion signal to source nodes in case of congestion is difficult resulting into further decrease in throughput.

SETM provides high per node throughput than in FETM because nodes report event without equally sharing the bandwidth among all event reporting nodes. CCF (Tien & Bajcsy, 2004) provides low per node throughput since packet service time at a node increases with density and also CCF does not bind service time with buffer occupancy as a result buffer is not optimally utilized.

6.4.2.1 Packet Delivery Ratio

The simulation scenario for multiple hop interference discussed in section 6.4.2 is used to evaluate the packet receive ratio. In Figure 6.13 it is shown that, for 50 nodes CCF provides high packet receive ratio since the density of event reporting nodes is less. However, due to interference and busy medium at high density, CCF fails to provide high packet receive ratio for 100 nodes. The packet receive ratio of FETM is above 95% under variable densities. This is because of the proposed congestion control mechanism that not only avoids but controls congestion by efficiently adjusting the reporting rate of congested nodes resulting into few packet drops. Moreover, schedule based packet forwarding provides an upper layer solution for packet drops due to interference.

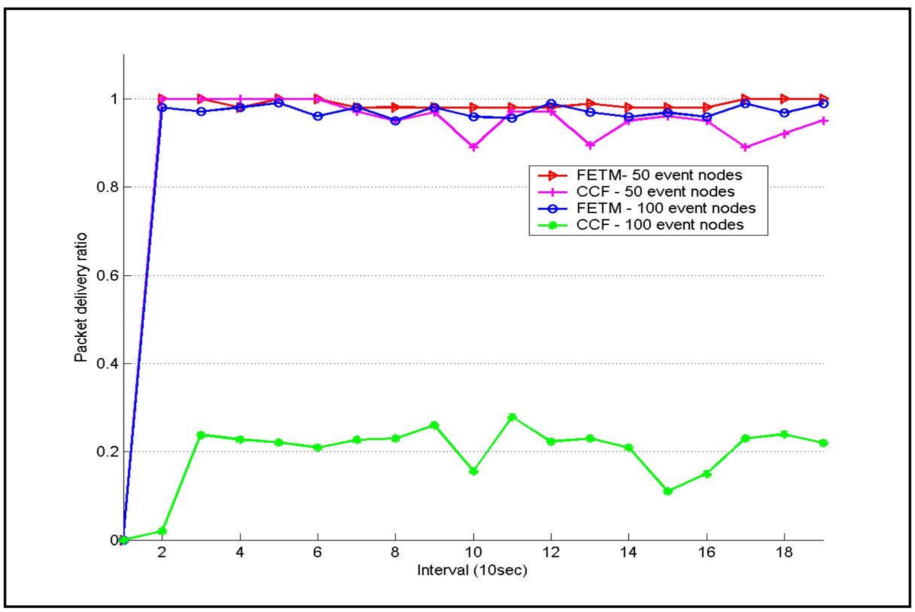

Figure 6.13 Packet delivery ratio of 50 and 100 event nodes randomly deployed and reporting event using Fair Event Transport Mode (FETM) and Congestion Control with Fairness (CCF).

6.4.2.2 Schedule Based vs. Jitter Based Packet Forwarding

The proposed schedule-based scheme, in this study, is compared with a simple jittered based forwarding scheme, when both applied at transport layer. In both these schemes, proposed congestion control mechanism based of packet delivery time and buffer size is used. In jittered forwarding version, routing nodes send reporting rate to their previous hop node instead of a schedule. In order to achieve fair per node throughput like CCF, each node forwards packets equal to its sub-tree size. Per node throughput observed at the actor for 100 randomly distributed event reporting nodes in an event region centered at coordinates (40,40), with an event diameter of 40m, is shown in Figure 6.14.

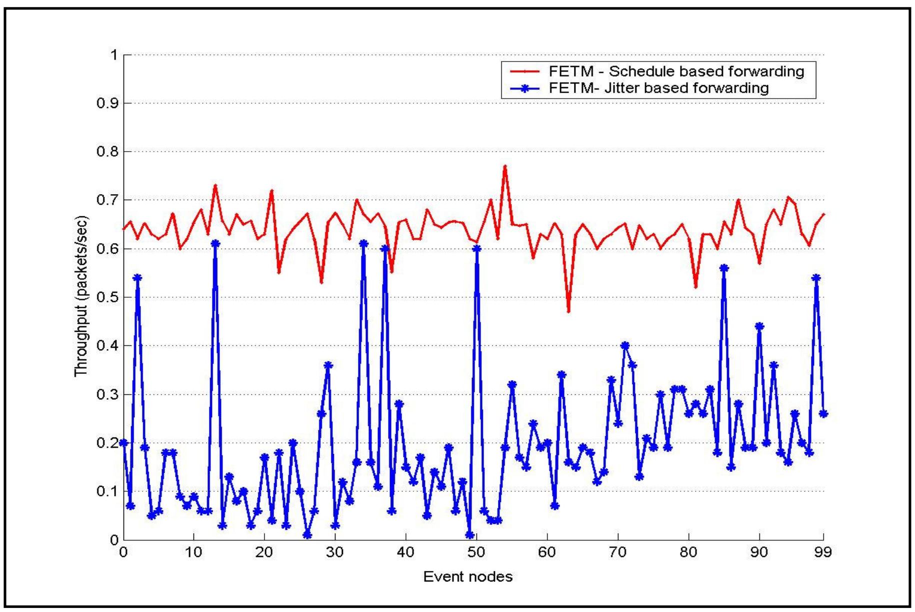

Figure 6.14 Per node throughput of 100 event nodes randomly deployed and reporting using Fair Event Transport Mode (FETM) with schedule and jittered based forwarding.

Separate simulations are conducted for 120sec of event reporting. In case of jittered forwarding due to high density, collisions and busy medium; the buffer of nodes start to overflow resulting into packet drops. However, by using scheduled transmissions packet delivery ratio increases resulting into fair and increased throughput (Figure 6.14).

6.4.4.3 Energy Consumption

The residual energy of a network, comprised of 100 and 150 event reporting nodes in 100x100m sensor field is shown in Figure 6.15. Initial energy of all the nodes is set to 0.1 Joules for this simulation.

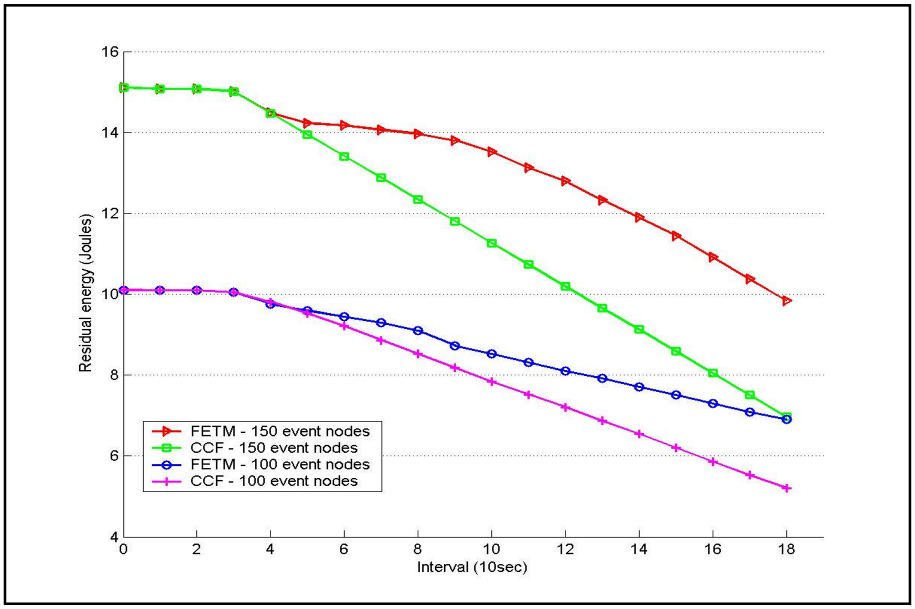

Figure 6.15 Residual energy 100 and 150 nodes network during 150sec of event reporting using Fair Event Transport Mode (FETM) and Congestion Control with Fairness (CCF).

Despite of additional scheduled packet transmission, FETM decreases the energy consumption. Since it handles congestions efficiently, it does not increase the reporting rate of nodes more than they can handle.

6.4.3 Prioritized Event Transport Mode (PETM)

PETM handles transport of more than one event according to their initial event reporting rates. In order to evaluate the performance of PETM, the node arrangement used in this study is shown in Figure 6.6. Two different simulations are conducted for two and four separate events. In the first simulation, the event reporting nodes report two different events E_1 and E_2, such that the initial reporting rate of E_2 is twice that of E_1. While in the second simulation, four different events E_1, E_2, E_3 and E_4 are reported by nodes such that the initial reporting rate of events E_2, E_3 and E_4 is twice, thrice and four times that of event E1 respectively. The events and the respective nodes reporting those events are shown in Tables 6.14 and 6.15.

Table 6.14 Event reporting nodes for event E_1 and E_2.

Event	Event reporting node ID
E_1	3,9,11,17,19
E_2	1,2,4,5,6,7,8,10,12,13,14,15,16,18,20

Table 6.15 Event reporting nodes for event E_1, E_2, E_3 and E_4

Event	Event reporting node ID
E_1	1,5,6,7,15
E_2	2,8,9,13,14
E_3	16,17,18,19,20
E_4	3,4,10,11,12

The per node throughput of event nodes at the actor which are reporting two and four events respectively are shown in Figures 6.16 and 6.17. Since CCF only considers node based fairness, it is unable to provide fair event reporting with respect to multiple event demands. Proposed transport modes includes PETM, which uses initial event reporting rate for rate allocation therefore nodes according to the event demand get a share of the bandwidth.

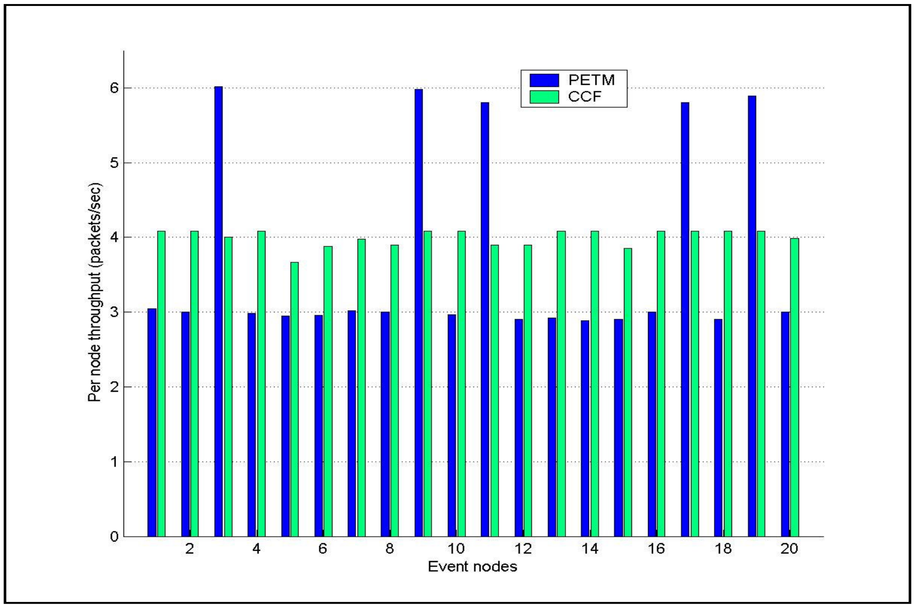

Figure 6.16 Per node throughput of 20 event nodes reporting two different events using Prioritized Event Transport Mode (PETM) and Congestion Control with Fairness (CCF).

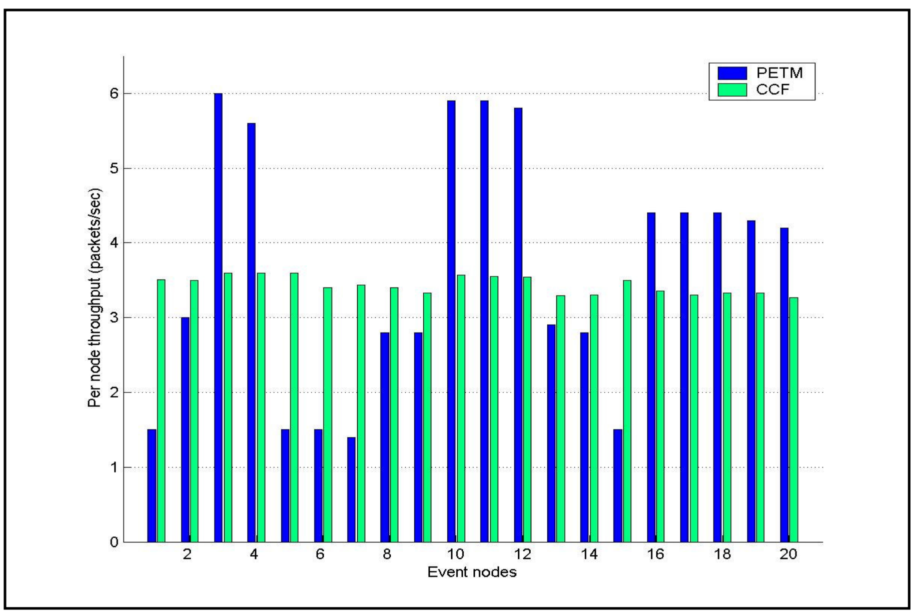

Figure 6.17 Per node throughput of 20 event nodes reporting four different events using Prioritized Event Transport Mode (PETM) and Congestion Control with Fairness (CCF).

The packet delivery ratio of the four events E_1, E_2, E_3 and E_4 using PETM is shown in Figure 6.18. Congestion has almost same affect on all event flows as the packet delivery ratio decreases, when congestion occurs. PETM effectively controls congestion and sustains a high packet delivery ratio for all the events.

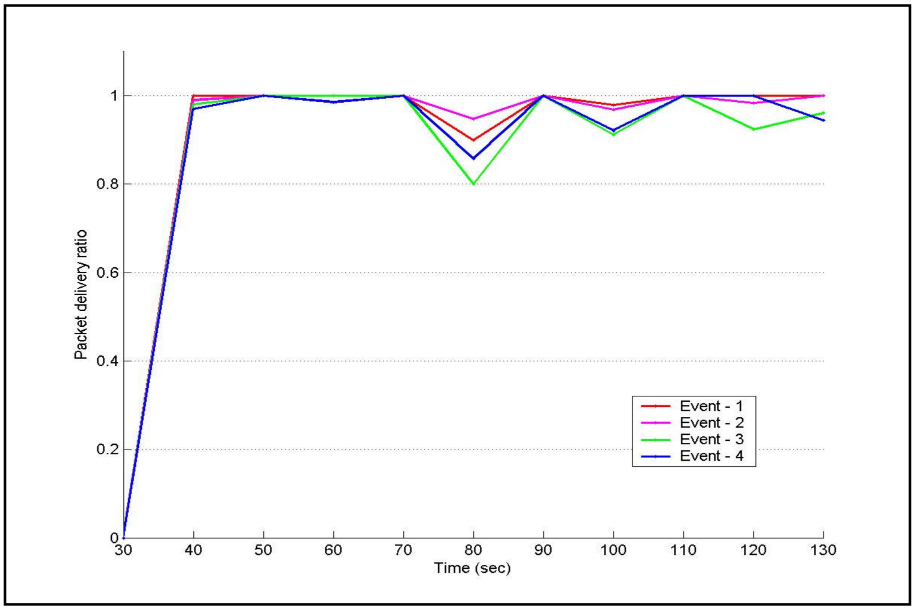

Figure 6.18 Packet delivery ratios of four different events, reporting events using Prioritized Event Transport Mode (PETM) and Congestion Control with Fairness (CCF).

CHAPTER SEVEN
REAL TIME EVENT TRANSPORT IN WIRELESS SENSOR AND ACTOR NETWORKS

7.1 Motivation

Wireless Sensor and Actor Networks (WSANs) are deployed to detect a variety of events. In case of time critical event information, the transport solution has to ensure that the event packets must reach the destination within certain time bound, while meeting the reliability requirements. The effectiveness of an action by the actors depends on the reliability of event information in terms of the magnitude and in-time delivery of an event (Culler, Estrin, & Srivastava, 2004).

Time bound transport is defined as *number of in-time packets received at the destination(s) to the required number of in-time packets for reliable event detection*. For time bound transport, a packet needs to be forwarded in such a manner that it should reach the destination before its deadline expires. The three basic things required for time bound transport are:

- A strategy for forwarding time critical information first
- Congestion detection
- Rate adjustment to avoid congestion while providing either application's defined reliability or maximum throughput.

The issue of time bound information delivery to the destination is new in WSANs (Gungor, & Akan; 2007). Some existing studies propose routing and cross layer protocol design solutions for time bound information delivery. Gungor, & Akan (2007), propose a reliable transport of time bound information to the destinations in WSANs. Their work is a destination-based transport solution in which the destination controls the reliability, reporting rate and congestion occurring in the network. However, destination based solution are slow to react to congestion and also such solutions are unable to achieve high throughputs (as discussed in sections 4.2.1 and 6.3).

In this study, two different solutions for time bound event transport in wireless sensor networks are examined. Simple sensors-to-Actor Real-time Event Transport protocol (SARET) and Time-bound Event Transport Mode (TETM).

- SARET protocol: The protocol uses in-network congestion detection based on buffer occupancy of nodes and an Additive Increase Multiplicative Decrease (AIMD) based rate adjustment scheme for multi-event prioritized transport. In order to achieve time bound event transport, nodes assign weight to individual event packets depending on their event priority, delay bound and packet delay. To achieve deadlines, packets with highest weight are forwarded first. In order to achieve required level of reliability, actors broadcast reliability status to event nodes for adjusting their reporting rates.
- TETM: This transport mode is a continuity of the proposed transport solution presented in section 4.1 and uses in-network based congestion mitigation and rate adjustment scheme with destination guided reliability achievement mechanism. By decoupling the destination's assistance for achieving reliability, the solution can then provide maximum system throughput. The proposed congestion control and schedule based rate adjustment are modified to fulfill real-time requirements (). A delay constraint based packet forwarding scheme is introduced for forwarding in-time packets with shortest remaining time first ().

7.2 Simple Sensors-to-Actors Reliable Event Transport (SARET) Protocol

SARET protocol is designed to provide reliable, delay-sensitive transport for prioritized multiple events in WSANs. Since SARET uses an easy to implement rate adjustment scheme based on AIMD and the congestion detection mechanism is based only on buffer occupancy of nodes, it is named as simple.

An automated WSAN architecture, with immobile sensor and actor nodes are considered in SARET's architecture. Nodes are randomly scattered within the network without any fixed topology, while the actor nodes are deployed in a way to provide maximum coverage. A hop-by-hop event delivery mechanism which is a modified version of Dynamic Source Routing (DSR) protocol (Johonson, & Maltz, 1996) is used in SARET. During the event transport, actors are capable of directly communicating and can exchange reliability information with each other. Based on hop distance, a node reports its events to the closet actor.

It is assumed in SARET, that all member nodes are capable of directly receiving their actor's broadcasts; as assumed by ESRT protocol (Akan & Akyildiz, 2005). In ESRT for achieving reliability, the destination periodically broadcasts a single reporting rate for all the event nodes. However, in SARET, the actors broadcast periodic signals indicating only reliability status at the actor not the required reporting rate to achieve reliability. The nodes adjust their reporting rate depending on their local buffer occupancy and the buffer status of their next hop nodes. This node based reporting rate selection method used by SARET allows different reporting rates to be maintained within the network, depending on the congestion status in a particular area. Therefore, in case of congestion, the network throughput is not much affected because the reporting rate of congested nodes is only decreased.

In SARET, reliability and delay constraint of an event are application defined parameters. The actors maintain intervals called *reliability interval* and at the end of each interval, the actors calculate observed reliability (OR) as ratio of total number of in-time packets received to the required number of in-time packets.

$$OR = \textit{in-time packets received / required number of in-time packets}$$

Since, more than one actor can receive the same event packets, actors coordinate with each other to maintain the overall network observed reliability ratio $OR_{NET} \approx 1$. Overall network observed reliability (OR_{NET}) ratio for an event is equal to the sum of individual OR ratios observed by different actors. If A_1 to A_n are actors observing the same event then overall network observed reliability will be:

$$OR_{NET} = OR_{A_1} + OR_{A_2} + \cdots + OR_{A_n} \tag{4}$$

7.2.1 Weight Assignment

SARET is aimed to serve multiple events according to their importance. Each node calculates packet weight on the basis of priority (P_i) of event E_i, delay bound (υ_i) and packet delay (τ). A packet with the highest weight is delivered to the routing layer for transmission to the next hop node. A weighted queue at the transport layer is implemented in SARET. All incoming event packets, as well as self generated event packets are placed in the queue.

In sensor networks, nodes nearer to the destination (sink/actor) deliver more packets to destination, when compared with nodes farther away from destination. Defining packet weight only on the basis of priority and delay bound can result into starvation. In order to avoid starvation, the total time spent by the packet in the network (τ) from the source node to the current node is considered. This includes all the delays i.e., waiting time in the transport queue and communication delay of the previously traversed nodes.

Packets spending more time in the network get higher weight than the fresh packets. Intuitively, packets originating from nodes which are away from the destination get an equal chance to reach the destination, on the basis of its weight. Hence, the weight W_i of a packet x belonging to the event E_i can be calculated as:

$$W_i^{\,x} = \frac{1}{\upsilon_i} + \left(\frac{P_i}{\sum_{j=1}^{k} P_j} \right) \times \tau \tag{5}$$

where k is the number of events being reported by the nodes. Weight is inversely proportional to the delay bound, i.e. weight is higher at small delay bounds but lower at large value of delay bounds. Similarly, priorities of the events are initialized by the application depending on their criticality. Critical events are represented by higher value of priority which increases the packet's weight.

7.2.2 Operation of Simple Sensor-to-Actors Reliable Event Transport

Nodes in SARET start reporting an event E_i at application defined initial reporting rate. A node can be relaying, as well as, sending its own event readings. Nodes transmit a packet only if there is no congestion at the node itself or at their next hop nodes. Nodes multiplicatively increase or decrease their reporting rate depending on locally detected congestion.

Reliability level is measured by the coordinated effort of actors after each interval, by using the equation 4 given above. At the end of each interval, the reporting rate is adjusted according to the required reliability level RR, as given below:

- $OR_x < RR_x$: If the network observed reliability for any event x is less than the required reliability for the event, actors broadcast the *increase reporting rate* message to the sensor nodes. Accordingly, member nodes of the actor respond by increasing their reporting rates. Nodes increase their reporting rate, only if their local and next hop node's buffer is not congested. If congestion is detected, then a node does not increase its reporting rate until congestion mitigates.

  Let ϕ_i^k be the reporting rate of a node i after k^{th} interval, then increased reporting rate for $(k+1)^{th}$ interval will be:

$$(k+1)^{th} = \alpha\phi_i^k \tag{6}$$

Where α is the multiplicative factor which is set as $1 < \alpha < 2$. The higher values of α help to achieve the reliability quicker. However, high values of α may unnecessarily increase the reporting rate beyond the required level causing congestion in the network and can affect the reliability of other events.

- $OR_x > RR_x$: If the network observed reliability for any event x is greater than the required reliability for the event, actors announce *decrease reporting rate* message to the sensor nodes at the end of reliability interval. Nodes decrease their reporting rate until the required reliability level is achieved. This ensures that the nodes do not over exert themselves by reporting at a rate greater than the required rate.

Let ϕ_i^k be the reporting rate of a node i after k^{th} interval. Then, decreased reporting rate for $(k+1)^{th}$ interval will be:

$$(k+1)^{th} = \beta\phi_i^k \tag{7}$$

Where β is multiplicative decrement factor which is set as $0 < \beta < 1$. The higher values of β can carefully and slowly regain the required reliability, as the reporting rate is decreased at a slower rate. However, for smaller value of β may unnecessarily decrease the reporting rate below required level causing decrease in observed reliability.

- $OR_x \approx RR_x$: In this case, network observed reliability for the event is approximately equal to its required reliability and the actors remain silent and sensor nodes keep reporting events at their current rate.

In order to achieve application based reliability, the algorithm used for adjusting the reporting rates of nodes is given below:

```
1. If LOCAL_BUFFER = LOW then
      /* Node's buffer is not congested */
2.    CALL SEND_CONGESTION_MSG(INCREASE)
      /* Send message to previous hop or child nodes for
         increasing reporting rate */

3.    If ACTOR_MSG = INCREASE and NEXT_HOP_MSG = INCREASE
      then
4.       NODE_RR = α × NODE_RR
         /* Node's reporting rate (NODE_RR) */

5.    Else if ACTOR_MSG = MAINTAIN and
      NEXT_HOP_MSG = INCREASE then
6.       NODE_RR = NODE_RR
          /* Same reporting rate used in previous interval */

7.    Else if ACTOR_MSG = DECREASE and
      NEXT_HOP_MSG = INCREASE then
8.       NODE_RR = β × NODE_RR

9.    Else
      /* Next hop nodes are congested therefore act according
      to   selected mitigation scheme*/
10.      CALL LOCAL_CONGESTION_MITIGATION
         OR
         CALL SOURCE_BASED_CONGESTION_MITIGATION
11.   End if

12. Else /* Node's buffer is congested */
13.   NODE_RR = NODE_RR / 2 /* Decrease node's reporting
rate
                                          to half*/
14.   CALL SEND_CONGESTION_MSG(DECREASE)
      /* Send message to previous hop nodes for  decreasing
         reporting rate */

15. End if
```

7.2.3 Congestion Control

In SARET, each node sends periodic beacons to their previous nodes about their local buffer condition. Additionally, next hop nodes predict for the upcoming traffic in the next interval by recording the reporting rate of previous hop nodes to avoid congestion. SARET implements in-network congestion control mechanism, so that the congestion information is not relayed to the actors.

Nodes maintain a list of all the previous hop nodes reporting events through them. Additionally, nodes maintain *congestion-intervals* that are much smaller than the *reliability interval*. This allows the nodes to immediately send congestion indication to the upstream nodes as it occurs.

7.2.3.1 Congestion Detection

In SARET, congestion is predicted on the basis of rate of change in buffer occupancy per congestion-interval and on the basis of event-traffic that a node will receive during the next reliability interval. The former is used to predict the congestion during next congestion-interval because the buffer occupancy increases almost linearly during a single congestion interval; as congestion intervals are small. Since reliability intervals are considerably large as compared to congestion intervals, the buffer occupancy is variable in reliability intervals. Therefore, the later is used to predict the congestion during the next reliability. The buffer occupancy during the next congestion-interval can be predicted by following a statistical approach using equation 1 and 2, in section 5.3.1.

A node in SARET, predicts the congestion for every reliability interval by calculating the event traffic that will pass through the node during next interval. Let ρ_i^k be the number of packets received by the i^{th} node during the k^{th} interval from one of its previous hop nodes. Then the total number of packets received by the i^{th} node during k^{th} interval from all of its n previous hop nodes are $\sum_{l=0}^{n} \rho_i^l$. If b_i^k is the buffer occupancy after the k^{th} interval of i^{th} node, then the predicted buffer occupancy for $(k+1)^{th}$ interval can be given as:

$$b_i^{k+1} = \frac{b_i^k}{\sum_{l=0}^{n} \rho_i^l} \times \left(\alpha \sum_{l=0}^{n} \rho_i^l \right) \tag{8}$$

Where α is the increment factor of reporting rate as described in section 7.2.2.

7.2.3.2 Congestion Mitigation

Two schemes for mitigation of congestion; local mitigation and source-based mitigation are presented, which are aimed at decreasing energy consumption and increasing throughput respectively. The details of both these schemes are given below:

- *Local Congestion Mitigation:* In this scheme, a node detecting congestion sends congestion signal to its previous hop nodes to decrease their reporting rate to half. Apart from this, the node also stops generating its own packets, until the buffer occupancy decreases to below of a threshold value. When a node gets a congestion message, it takes a localized decision to further propagate the message to its previous hop nodes or to simply decrease its reporting rate, depending on its local buffer occupancy. If required, the congestion signal can reach the source nodes to decrease their reporting nodes. Once buffer size of congested nodes decrease to below of a threshold value, they send congestion release message to their previous hop nodes, which then increase their reporting rates.

  This scheme immediately decreases packet losses due to buffer overflow because all nodes from the congested node to the source nodes can decrease their reporting rates. However, the throughput of the system is affected as the reporting rate of all previous nodes is decreased to half.
- *Source-based Congestion Mitigation:* Source-based scheme can only mitigate the congestion by reducing the reporting rate of event-generating nodes rather than blocking event data at intermediate nodes. When an intermediate node is

congested, it simply drops packets and informs its upstream nodes to reduce their rate. Meanwhile, intermediate nodes continue to drop and relay packets until its buffer size decreases to below of a threshold value. Reporting rate of event-generating nodes is increased upon the reception of an increase message from the previously congested node. This scheme increases throughput of the system, as the reporting rate of only particular event-reporting nodes is decreased as compared to all the upstream nodes in local mitigation scheme. However, it results into more energy losses, as packets are dropped until congestion information is relayed to particular event-generating nodes.

7.2.4 Simulation Results for SARET Protocol

The performance of SARET protocol is evaluated in terms of achieving reliability, congestion mitigation and packet drop ratio. The example scenario of wireless sensor and actor network consists of 100 sensor nodes deployed randomly in a field of 100 x 100m. The configuration parameters for the simulations are summarized in Table 7.1.

Table 7.1 Simulation parameters.

Transport Layer	SARET
Network Layer	dynamic source routing
MAC Layer	802.11
Propagation model	Two-ray reflection
Deployment	Random
Packet length	30 bytes
IFQ Length	65 Packets
Transmit Power	0.660 W (fixed)
Receive Power	0.395 W
Radio Range	20m
Congestion interval	2 sec
Actor interval	10 sec

7.2.4.1 Reporting Rate vs In-Time Packet Delivery

With the increasing reporting rate of nodes, packet delivery ratio increases. However, after a certain maximum reporting rate, congestion starts to occur and time constraint of packets starts to exceed. At this stage, increasing the reporting rate considerably decreases the in-time delivery of packets. This is shown in Figure 7.1, where a single event with time constraint equal to the reliability interval (10 sec) is reported at different reporting rates.

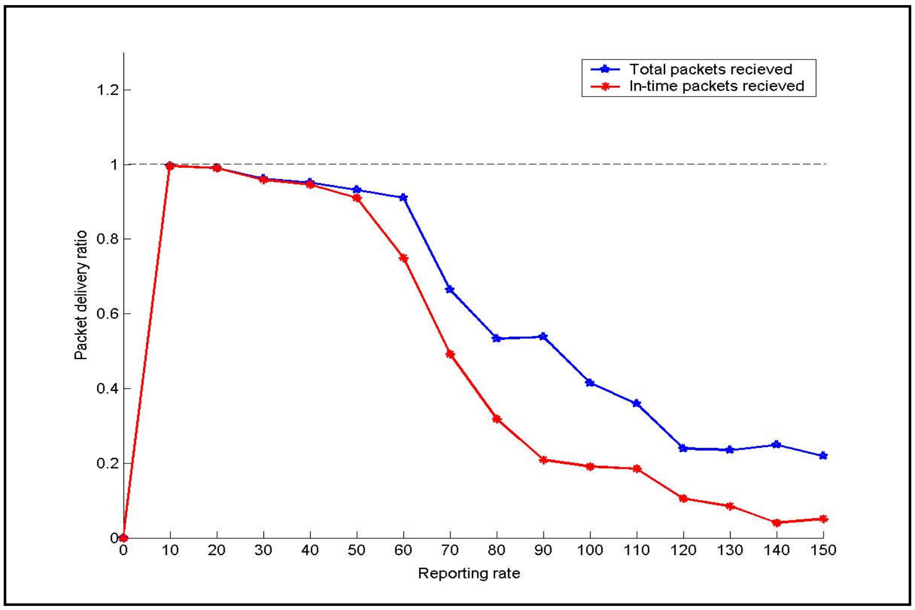

Figure 7.1 Affect of increasing reporting rate: 20 event-reporting nodes, event region centered at (75,75) are reporting to a single actor at (50,50).

7.2.4.2 Achieving Reliability

If the observed network reliability is either less than or greater than required reliability level, reporting rate of nodes is adjusted. In these simulations, 20 event nodes start reporting the same event with an initial application defined reporting rate (1 packet per second per event node) and increase their reporting rate on actor's messages. The required reporting rate to be observed at the actor is 800 packets per reliability interval. The affect of increasing α for a single event, is shown in Figure 7.2.

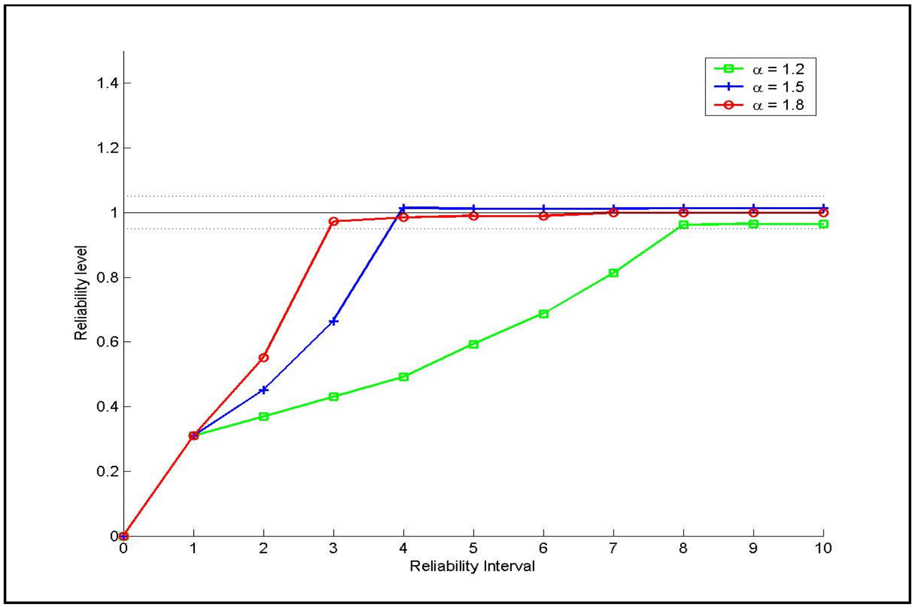

Figure 7.2 Low reliability scenarios; an event is reported by 25 nodes while using different values of α to achieve application defined reliability.

Reliability is achieved in 3^{rd} interval for α = 1.8, while reliability in cases of α = 1.5 and α = 1.2 is achieved in 4^{th} and 9^{th} interval respectively (Figure 7.2). Therefore, for higher values of α reliability is achieved quickly as compared to lower values.

If the reliability level is above required level, then SARET decreases the reporting rate slowly. This is shown in Figure 7.3. After 30 seconds of event reporting by 25 nodes, another 15 nodes join the event reporting with initial application defined reporting rate.

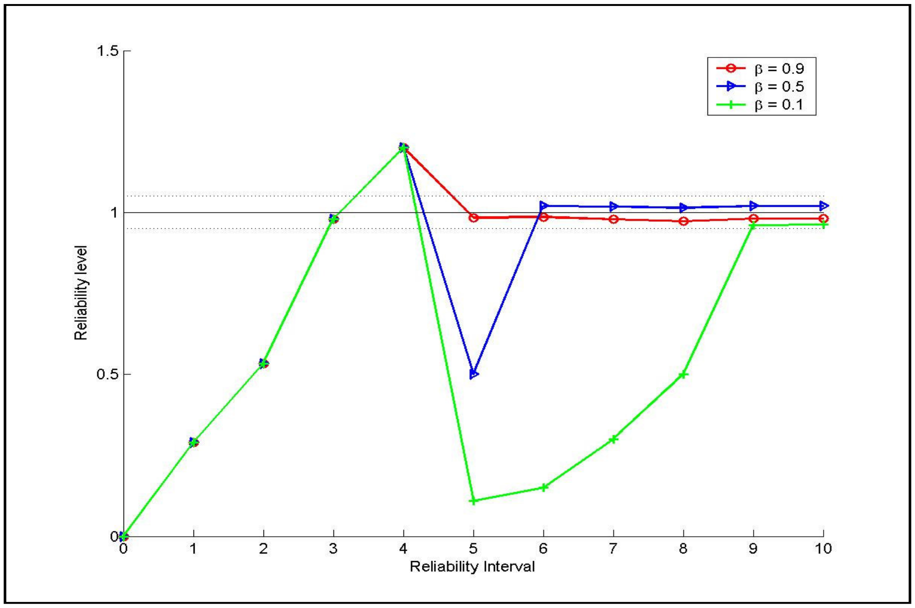

Figure 7.3 High reliability scenario; event nodes using different values of β adjust reporting rates to regain application defined reliability.

For higher values of β reliability is quickly re-achieved but for smaller value of β (such as 0.1), the reporting rate is decreased suddenly, resulting into low reliability (Figure 7.3). Depending on the total number of event nodes and the required reporting rate, an actor can initially broadcast the values of α and β to event nodes on event occurrence.

7.2.4.3 Congestion Control

The affect of congestion on in-time event transport, is shown in Figure 7.4, where the in-time event transport decreases considerably as congestion occurs. In this simulation scenario after achieving reliability, 10 additional nodes within the event region suddenly start reporting a different event, with very high reporting rate, resulting into congestion, which survives for approximately interval length. The congestion affects approximately 75 percent of event reporting nodes while other event reporting nodes are not affected by congestion.

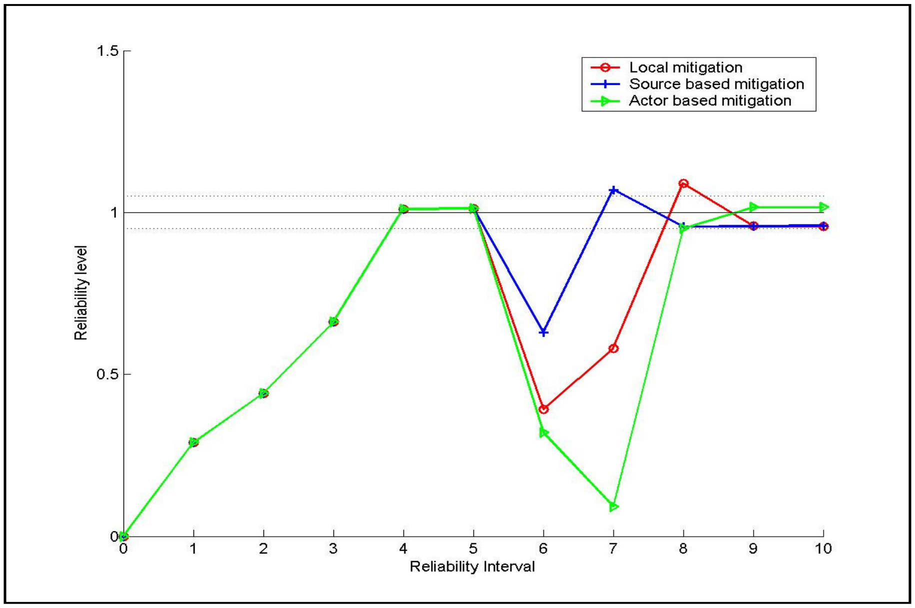

Figure 7.4 Comparison of local and source based congestion control in Simple sensors-to-Actor Event Transport (SARET) protocol with actor based congestion control ($\alpha = 1.8$ and $\beta = 0.9$).

Three implementations of congestion control mechanism are shown in Figure 7.4. Since the reporting rate of all the event nodes is not decreased in case of congestion, local and source-based congestion mitigation schemes used by SARET perform better. Also, the use of *congestion-intervals* by SARET which are much smaller than reliability intervals helps to detect congestion is detected earlier. On the other hand, actor based congestion scheme (similar to ESRT) detects congestion after the end of reliability interval resulting into late adjustment of reporting rate.

7.2.4.4 Packet Drop Ratio

The ratio of packet drops to the total number of in-time packets received during reliability intervals is shown in Figure 7.5. The simulation settings discussed in previous section are used to observe packet drop ratio.

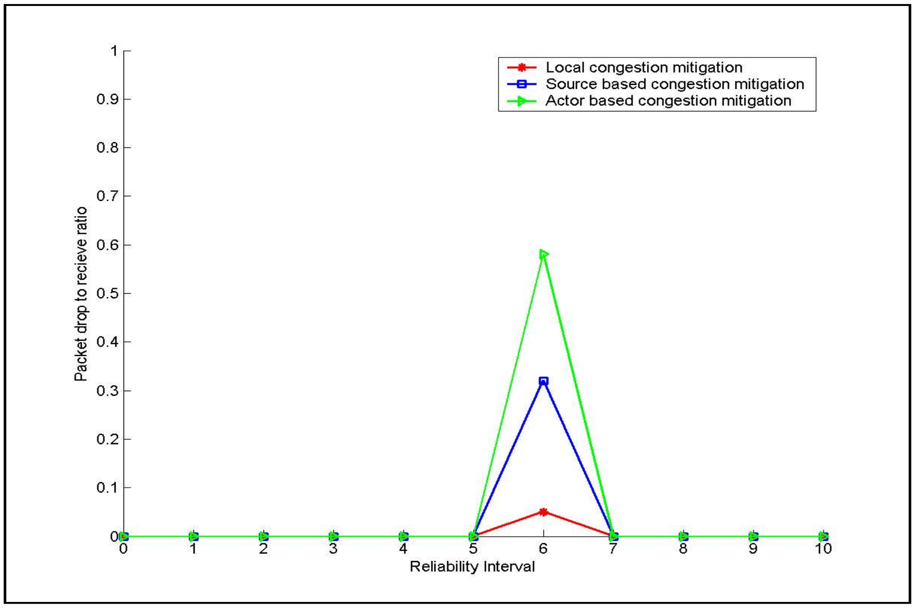

Figure 7.5 Ratio of packets dropped due to buffer overflow to total number of in-time packets received using different congestion mitigation schemes.

Since both congested node and previous hop nodes decrease their reporting rates immediately, local congestion mitigation scheme mitigates congestion more quickly and avoids packet drops. Source-based congestion mitigation scheme, results into more packet drops because congestion is mitigated by event-reporting nodes not by intermediate nodes. In this scheme, considerable numbers of packets are dropped until congestion message is received by an event reporting node. Actor based congestion scheme reacts poorly because congestion is detected after interval length resulting into more packet drops.

7.3 Time-bound Event Transport Mode (TETM)

TETM is built on top of the proposed congestion control and schedule based packet forwarding scheme. In order to provide time-bound transport, the packets are forwarded depending on the remaining time-bound of packet. The proposed schedule based scheme (discussed in chapter 5) is modified to meet real-time requirements.

The basic problem with SARET protocol is that, the rate adjustment scheme is based on fixed increment and decrement factors, which take into account neither the number of event reporting nodes nor the network conditions. Selecting a small value of increment factor can take a lot of time for achieving reliability, while selecting a large value of increment factor can result into immediate congestion on event occurrence.

In SARET, the reporting rate of nodes is not adjusted according to network conditions, but is decreased to half in case of congestion by each congested node; this decreases the system throughput. The congestion control scheme only considers buffer occupancy which increases slowly as compared to the actual congestion on the channel (Hu, & et al, 2005; Wan, Campbell, & Krishnamurthy, 2005). Therefore, congestion is generally detected once the buffer of a node overflows. In this situation, sending congestion control signal to previous hop nodes (which may also be congested) results in the drop of congestion signals.

The event, data and schedule headers in TETM are similar to the other proposed transport modes (shown in Tables 6.3, 6.4 and 6.5). However, the data and schedule headers contain additional fields "*time remaining*" and "*minimum hop delivery time*", respectively. The time remaining field contains the time after which the packet's time bound expires. Since the information is not useful or valid to the destination as the time bound expires, the packet is dropped by the node.

TETM forwards packets depending on the packet remaining time. Upon receiving a packet, a node updates the remaining time of the packet and places it in the transport queue. The transport queue is maintained in such a way, that the packet with the smallest time remaining is at the front of the queue. The minimum amount of time required for the transmission of a packet from one hop to the other can be named as "*minimum hop delivery time* (μ)", while "*minimum packet delivery time*" is the time required for the transportation of the event packet from the current node to the destination under ideal conditions.

Nodes can be at multiple hop distance from the destination, therefore for packets with remaining time smaller than minimum packet delivery time are dropped by intermediate nodes in TETM. This helps to reduce energy consumption and lowers the load on the network. Let d_i be the hop distance of a node i from the destination and μ be the minimum hop delivery time then the minimum packet delivery time for a packet at node i will be $d_i \times \mu$.

Nodes can use a constant value of μ which can be mirrored to match the minimum packet delivery time between a node and its next hop node under ideal conditions. However, hop delivery time can vary at each hop depending on the local network condition in terms of load at the link and congestion status. In order to have more precise results, each node calculates the minimum hop delivery time (v) and passes it to their previous hop nodes in the schedule packet. The pervious hop node further sends to its child nodes the average of its calculated v and the received v. Hence, each node uses the average minimum hop delivery time between the current node and the destination for the value of μ in the next data interval.

The slot allocation and slot length calculation for TETM is similar to simple event transport mode (SETM), as it is assumed, that general delay constraint information is required at the destination, while per node throughput is not considered. The reliability definition for TETM is based on the number of in-time packets received by the destination to the required application's defined number; similar to SARET.

Proposed congestion control scheme for congestion mitigation is used in TETM and the operation of TETM is also based on consecutive schedule and data intervals. In real-time event transport, schedule packets insert an unnecessary delay on the transport of event information. Therefore, the working of schedule based packet forwarding scheme is modified and overlapping schedule and data intervals are used, as shown in figure 7.6.

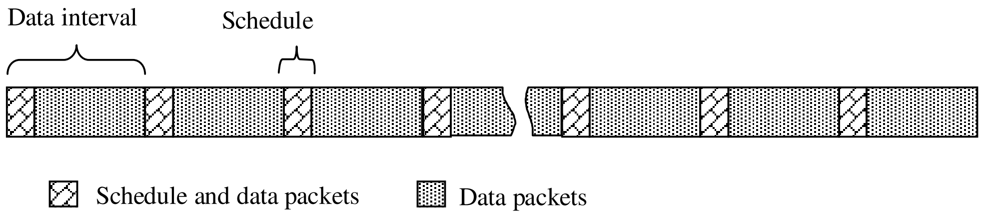

Figure 7.6 Overlapping data and schedule intervals maintained by nodes in time-bound even transport.

7.3.1 Simulation Results for TETM

The performance of proposed Time-bound Event Transport Mode (TETM) is observed using network simulator NS-2. The simulation scenario is comprised of a wireless sensor network, with 100 sensor nodes randomly deployed in a 100 x 100 m field. Minimum hop packet forwarding is used at the routing layer. Length of data interval and schedule interval is 5 and 1 seconds, respectively. The time-bound on each event packet is 2 seconds.

The performance of TETM is evaluated in terms of packet delivery ratio, throughput and energy consumption. The efficiency of TETM is compared with Simple sensors to Actors Real-Time Transport (SARET) protocol. For SARET, increment factor of 1.2 and a decrement factor of 0.9 are used. Different node arrangements are used to precisely evaluate the performance of TETM, which are given below.

- Sparse node arrangement: 20 nodes are deployed in an event region centered at coordinates (50, 50), with an event radius of 40 meters while the actor is at coordinates (90, 90). The nodes are arranged so that five first hop nodes having four child nodes each, report event to the actor.
- Low interference: In this scenario, interference is decreased by arranging nodes so that each node has no neighbor node expect for a next or previous hop node; as shown in Figure 7.7. The nodes are arranged in 9 hops such that each node only faces single hop interference.

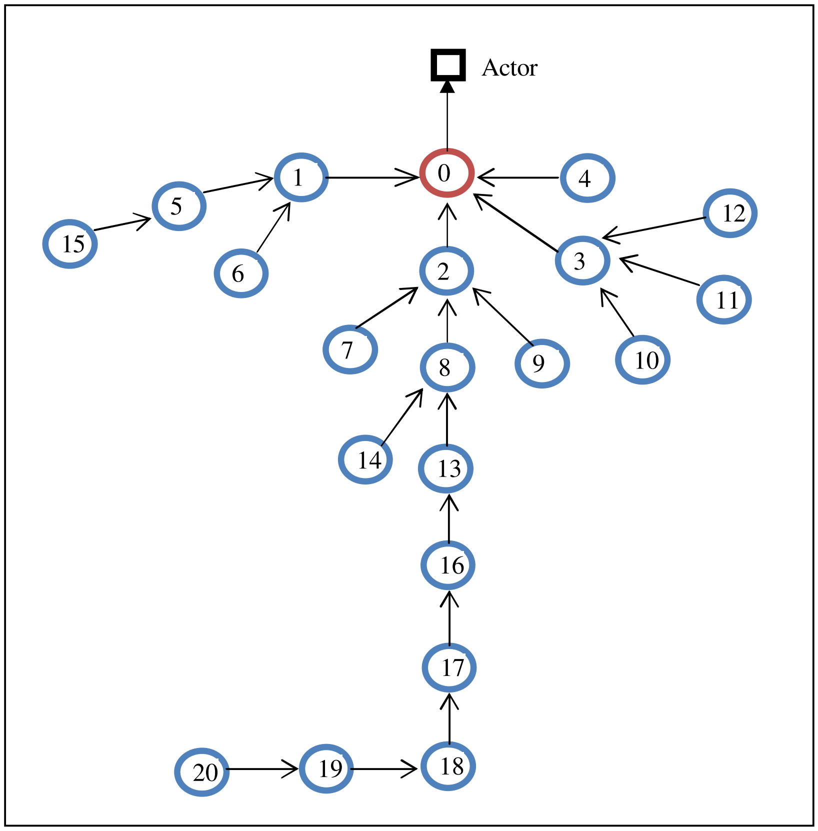

Figure 7.7 Arrangement of 21 nodes in a 100x100m sensor field with a maximum hop distance of 9 hops.

- Random node arrangement: In this scenario, 50 nodes are randomly deployed in an event region centered at coordinates (40, 40), with event radius of 20 meters while the actor is at coordinates (90, 90). The nodes are arranged in a single flow with only single first hop node.
- High node density: In this scenario, 100 event nodes are randomly arranged in an event region centered at coordinates (40, 40), with an event radius of 20 meters. Simulations are conducted for single and multiple flows with one and three first hop nodes, respectively.

7.3.1.1 Sparse Node Arrangement

For sparse node arrangement, the in-time packet delivery ratio is shown in Figure 7.8.

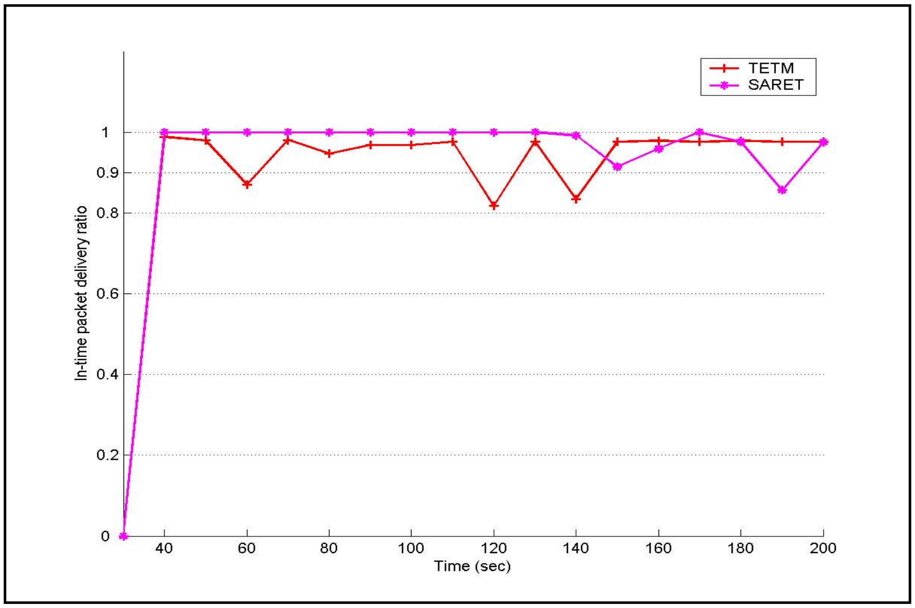

Figure 7.8 In-time packet delivery ratio of 20 event reporting nodes sparsely arranged, using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

Since the reporting rate of nodes is low due to small increment factor, SARET initially provides high packet delivery ratio. TETM also provides high packet delivery ratio but a decrease in the packet delivery ratio at different time instances is observed. This is because of congestion occurring at various first hop nodes (Figure 7.8). The average per node throughput for sparse node arrangement is shown in Figure 7.9.

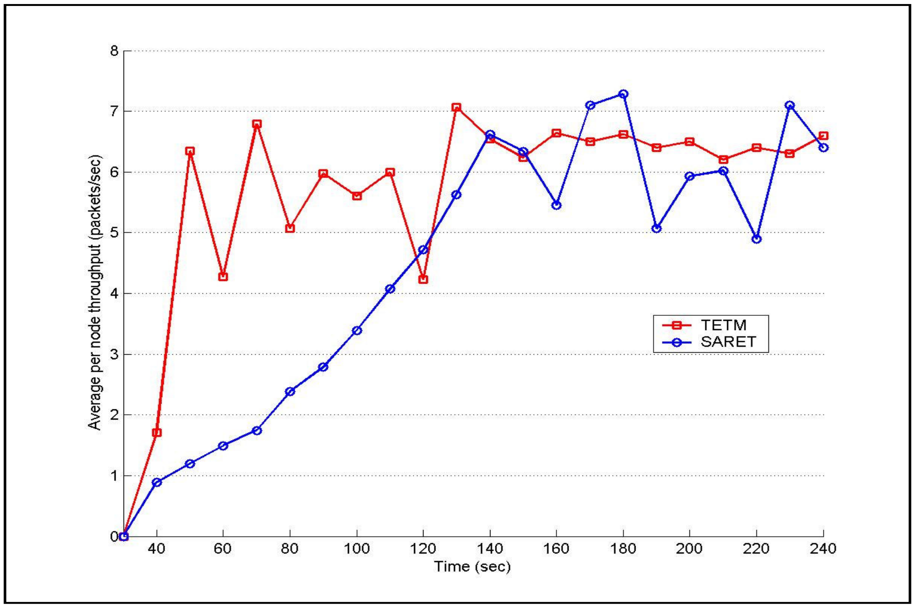

Figure 7.9 Average per node throughput of 20 event reporting nodes sparsely arranged, using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

The throughput of SARET protocol is random once the reporting rate increases and congestion occurs (Figure 7.9), still SARET provides high throughput. TETM on the other hand, depending on the channel conditions adjusts reporting rate of the nodes and achieves high throughput immediately.

7.3.1.2 Low Interference

In this case, nodes are arranged to provide lesser interference and to achieve high packet delivery ratio and more throughput. TETM provides high packet delivery ratio and throughput, as shown in Figures 7.10 and 7.11.

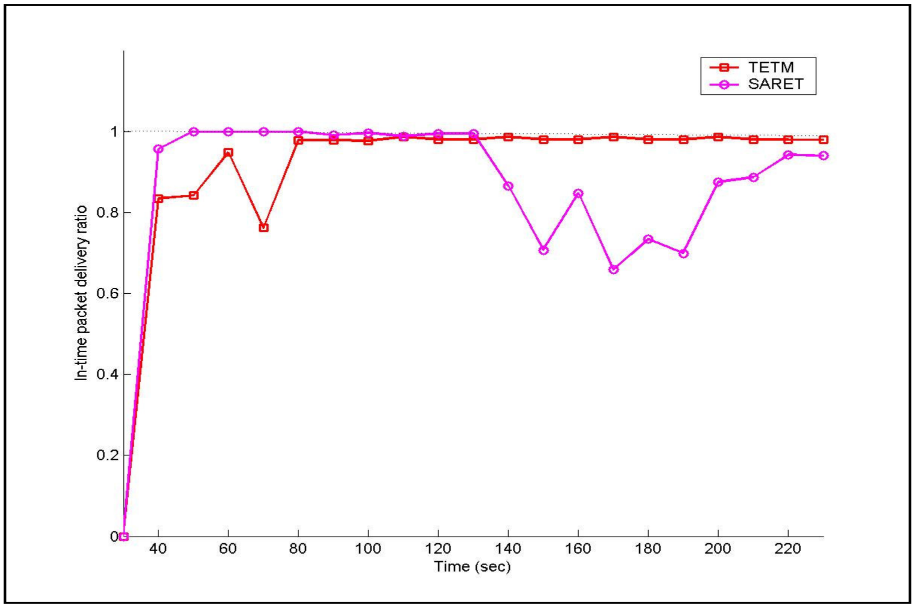

Figure 7.10 In-time packet delivery ratio of 20 event reporting nodes (arranged to provide less interference) using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

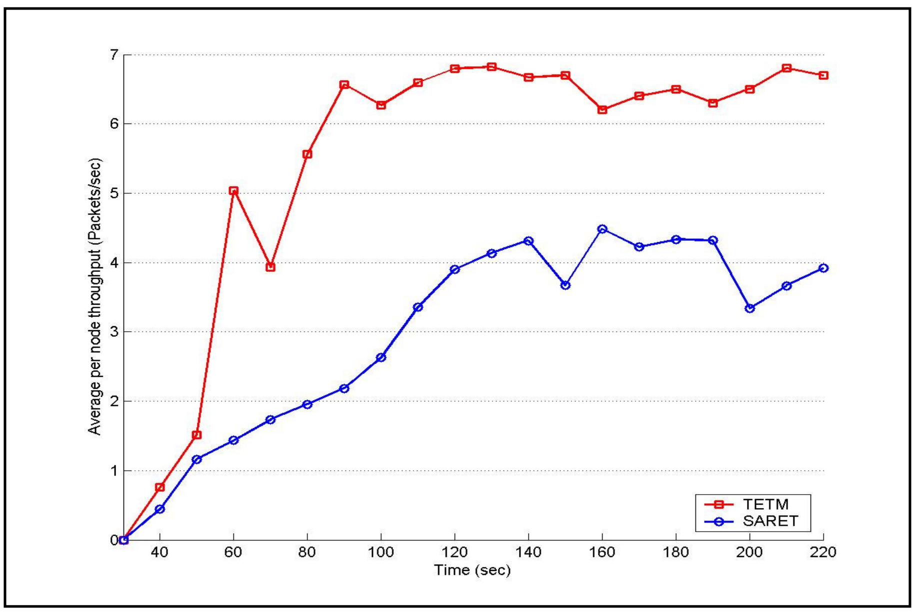

Figure 7.11 Average per node throughput of 20 event reporting nodes (arranged to provide less interference) using Time-bound Event Transport Mode (TETM) and Simple sensors to Actors Real-time Event Transport (SARET).

In case of SARET, the initial in-time packet delivery ratio is high due to small reporting rates and no congestion (Figure 7.10). Once congestion occurs sending congestion signals to source nodes which are at multiple hop distance are difficult. The reporting rate in SARET is not adjusted according to channel conditions; therefore a small increment factor can increase the reporting rate considerably, resulting again into congestion. The energy consumption of both the schemes is similar as shown in Figure 7.12.

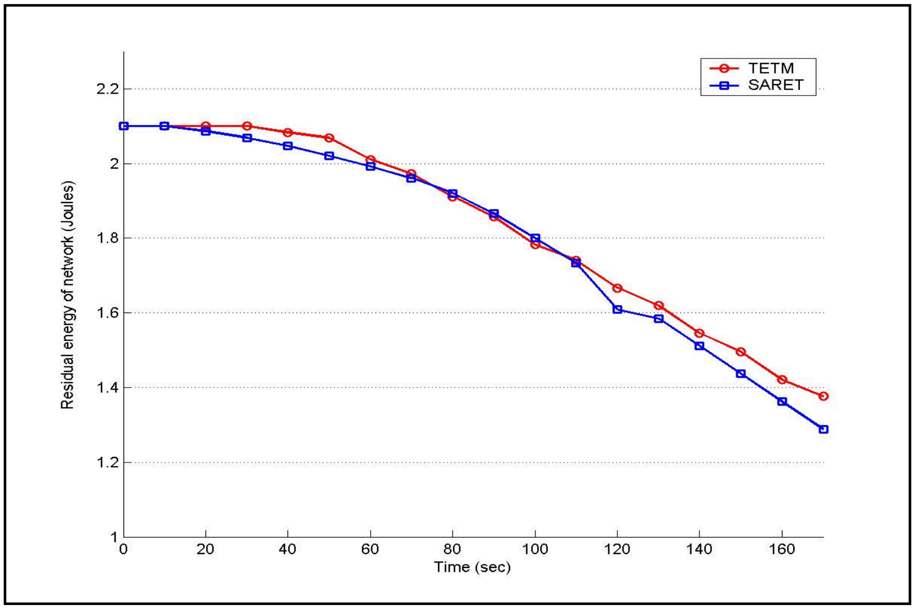

Figure 7.12 Residual energy of 21 nodes network with 20 event nodes, reporting an event using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET). Initial energy of each node in the network is 0.1 joules.

TETM provides higher throughput and much less packet drops, the effectiveness of TETM is evident from Figure 7.13, in which the ratio of throughput observed over energy utilized is shown. The higher the throughput and lesser the energy utilized in an interval (10 sec) the greater will be the output. Hence, in TETM, energy of nodes is efficiently utilized while providing high throughput.

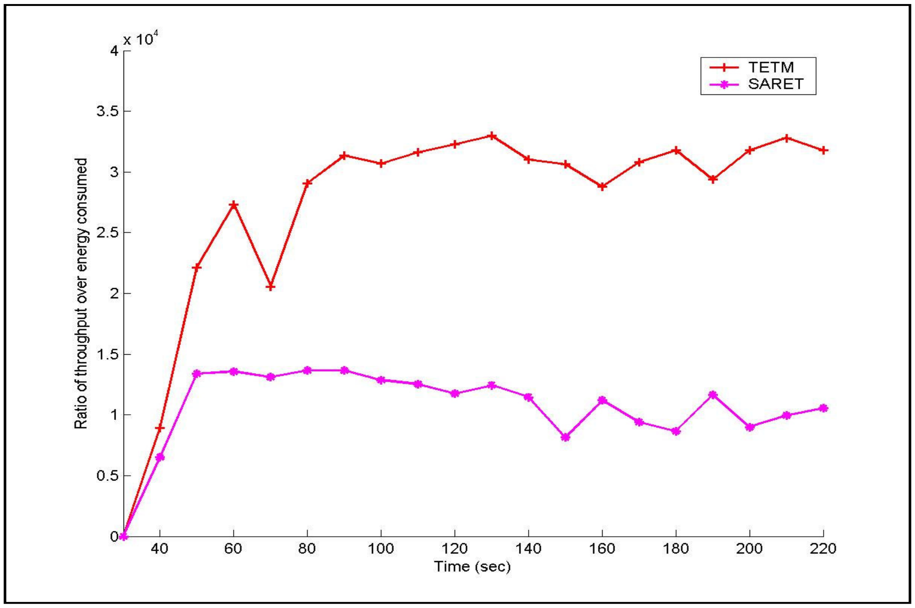

Figure 7.13 Ratio of throughput over energy consumed by 20 event nodes reporting an event using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

Both TETM and SARET are capable of achieving application defined throughput. In order to illustrate this fact, an application defined throughput of 600 packets per 10 seconds interval is used. The number of packets received at the destination is shown in Figure 7.14. For SARET, an increment factor of 1.2 and decrement factor of 0.9 is used in this simulation.

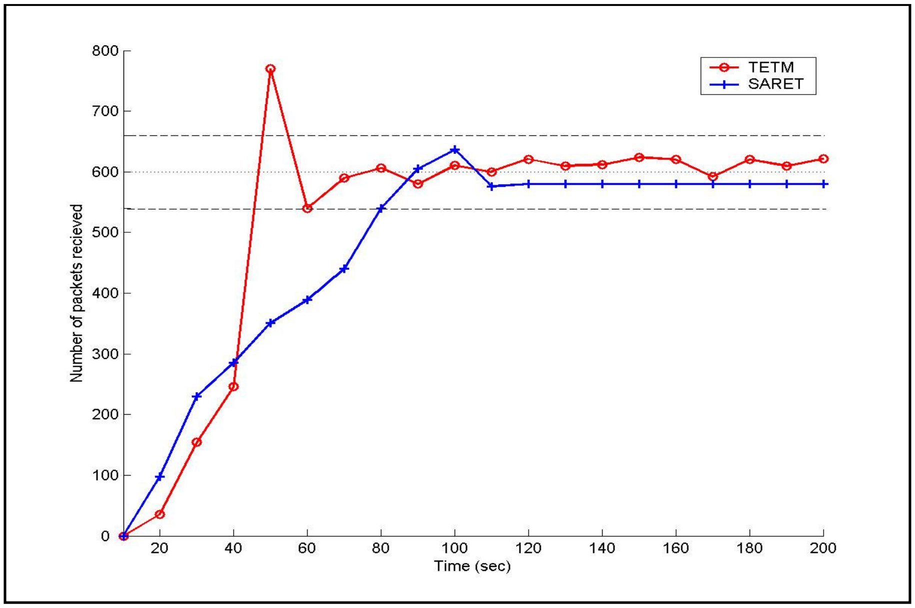

Figure 7.14 Number of packets received from 20 event reporting nodes using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET); required number of packets per 10 sec = 600.

TETM does not use any fixed increment and decrement factors, instead reporting rate of nodes is adjusted according to packet delivery time and buffer size of nodes. As a result, TETM achieves required throughput earlier than SARET. The energy consumption of SARET and TETM, while achieving required throughput of 600 packets per 10 seconds is shown in Figure 7.15.

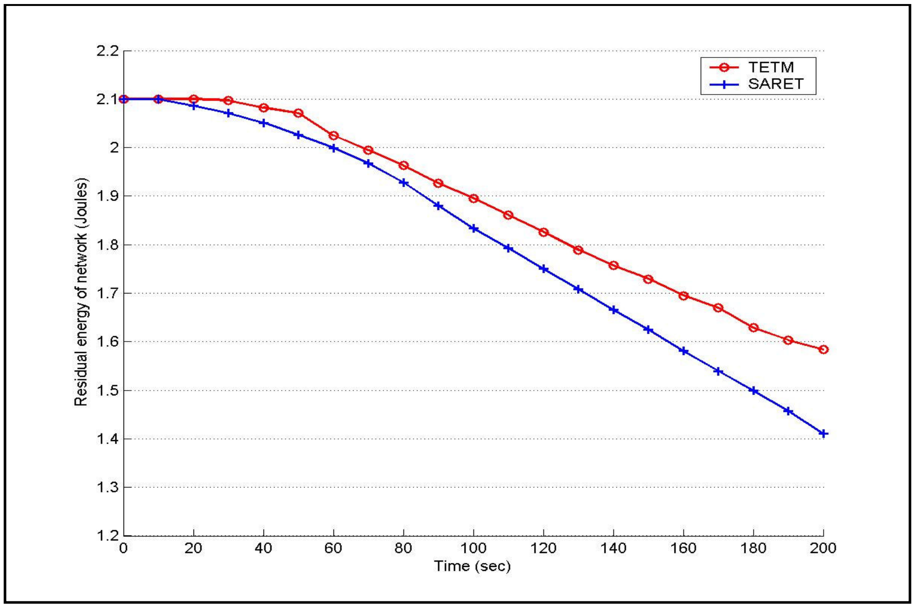

Figure 7.15 Residual energy of 21 nodes network with 20 event nodes, while achieving throughput of 600 packets per interval, using Time-bound Event Transport Mode (TETM) and Simple sensors-to-|Actors Real-time Event Transport (SARET).

Since TETM provides high throughput while utilizing less energy, TETM achieves the required reliability with less consumption than SARET (Figure 7.15).

7.3.1.3 Random Node Arrangement

In the random node arrangement scheme, considerably large numbers of nodes (50) are reporting an event through a single hop node and the density in the event region is approximately 25 nodes. The sudden impulse of event information results into congestion, which decreases the packet delivery ratio and throughput of SARET protocol, as shown in Figure 7.16 and 7.17.

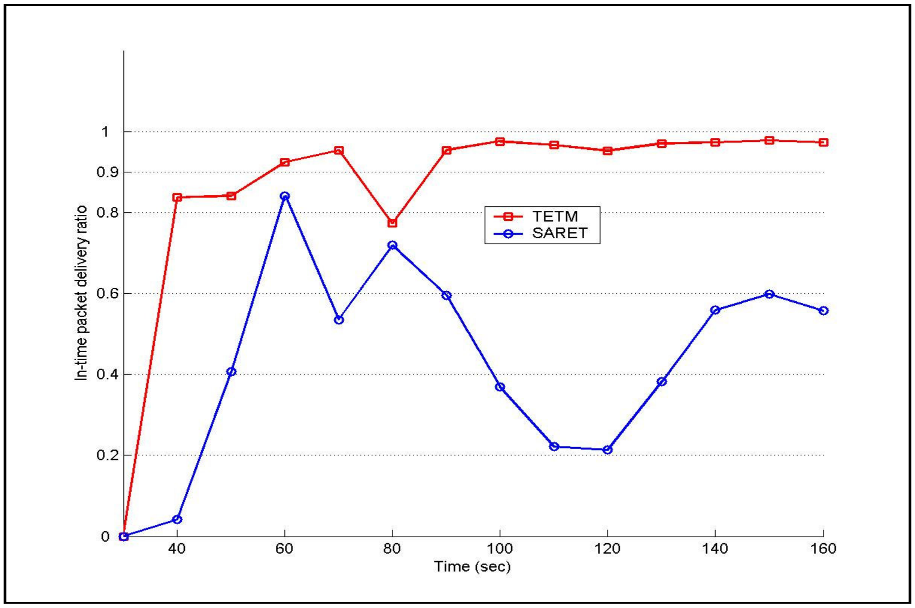

Figure 7.16 In-time packet delivery ratio of 50 event nodes randomly deployed in a single flow and reporting an event using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

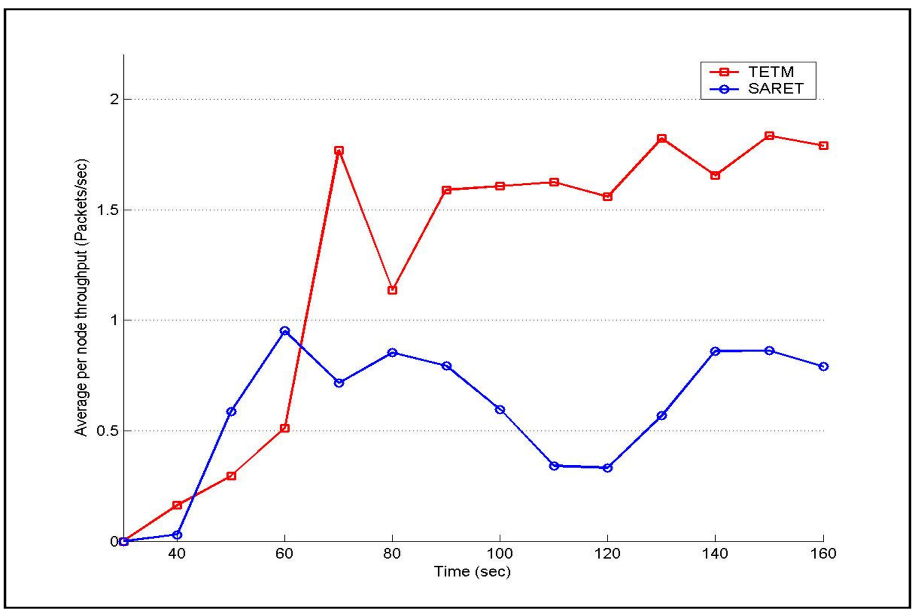

Figure 7.17 Average per node throughput of 50 event nodes randomly deployed in a single flow and reporting an event using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

The affect of interference increases in random node arrangement. As a result in case of SARET, congestion occurs more frequently at different nodes. SARET is unable to handle congestion in considerably dense scenarios. TETM provides low throughput initially (Figure 7.17), due to small slot length (on event occurrence 0.1 sec). Also, the channel becomes suddenly busy on event impulse resulting in increase in slot length but the packet drop ratio in case of TETM is high, showing a very few packet drops (Figure 7.16).

The residual energy of the network during 110 seconds of event reporting and the ratio of throughput over energy consumed by TETM and SARET are shown in Figures 7.19 and 7.20.

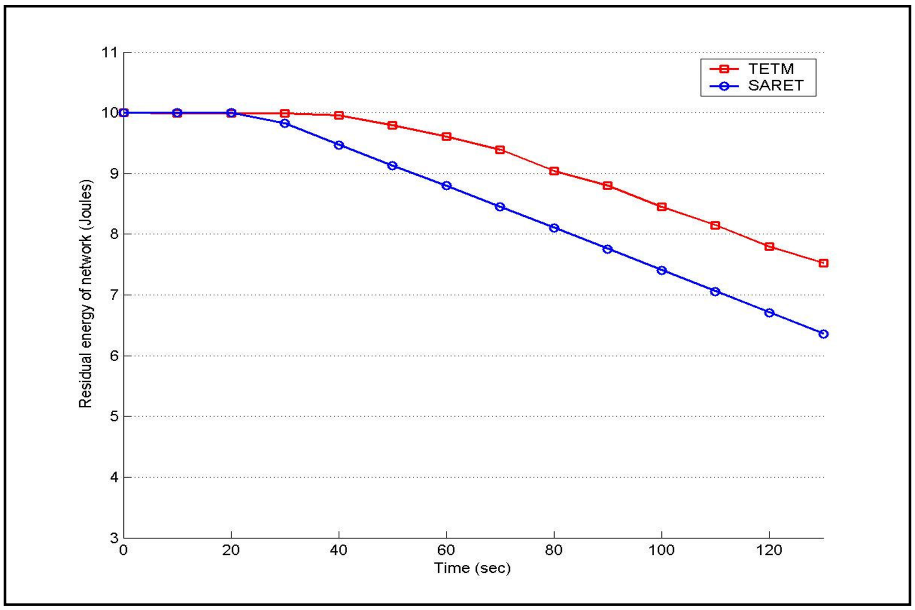

Figure 7.19 Residual energy of a 100 node network with 50 event nodes, reporting an event using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET). Initial energy of each node is 0.1 joules.

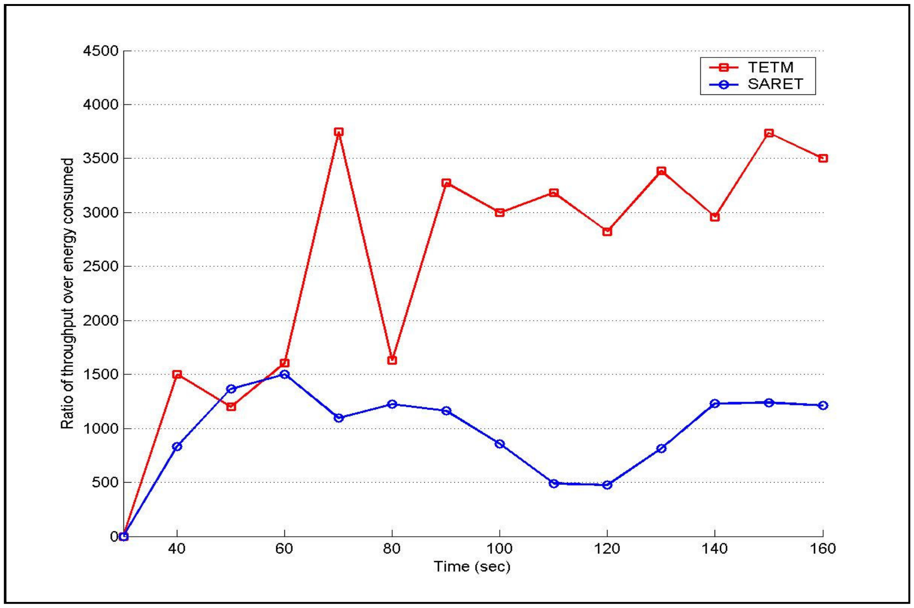

Figure 7.20 Ratio of throughput over energy consumed by 50 event nodes randomly deployed in a single flow and reporting an event using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

Since congestion is avoided and reporting rate of nodes is increased to provide high throughput under local channel conditions, TETM achieves more throughput while utilizing less energy (Figures 7.19 and 7.20).

The average packet delivery delay observed during event transport, is shown in Figure 7.21.

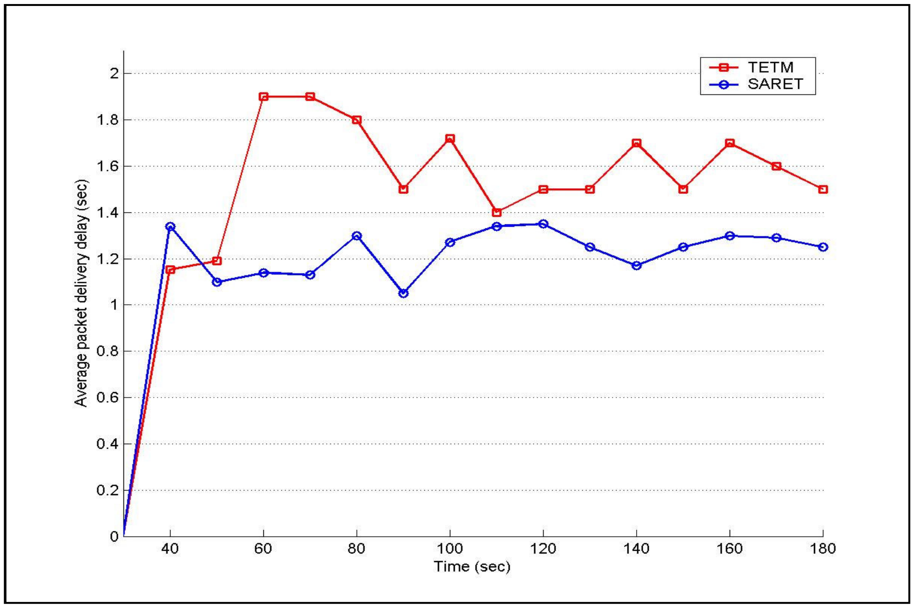

Figure 7.21 Average packet delivery delay observed by 50 event nodes randomly deployed in a single flow and reporting an event using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

An interesting fact shown in Figure 7.21 is that packets arrive late in case of TETM, as compared to SARET. The reason for this delay is the combined affect of both high throughput and shortest remaining deadline forwarding policy used in TETM. This fact is further elaborated in Figure 7.22 in which, the ratio of throughput over average packet delivery delay is shown.

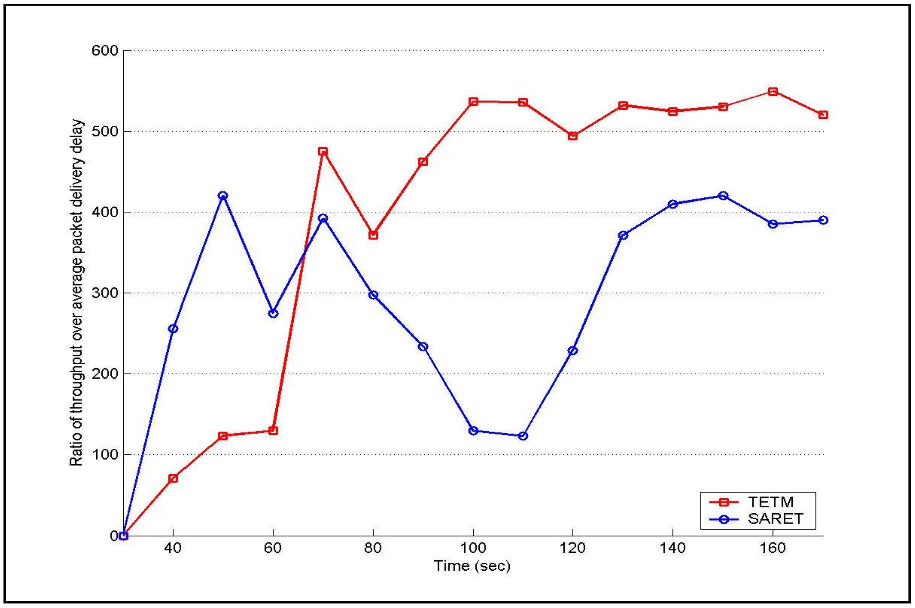

Figure 7.22 Ratio of throughput over average delivery delay by 50 nodes in a single flow and reporting using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

The higher the throughput or shorter the average delivery delay, higher will be the output. The output of TETM is higher than SARET, due to the greater throughput of TETM (Figure 7.22).

7.3.1.4 High Node Density

In case of node arrangement with high density, 100 event nodes are used for event reporting, with an approximate node density of 50 nodes. The in-time packet delivery ratio of event nodes arranged in a single flow and multiple flows is shown in Figure 7.23.

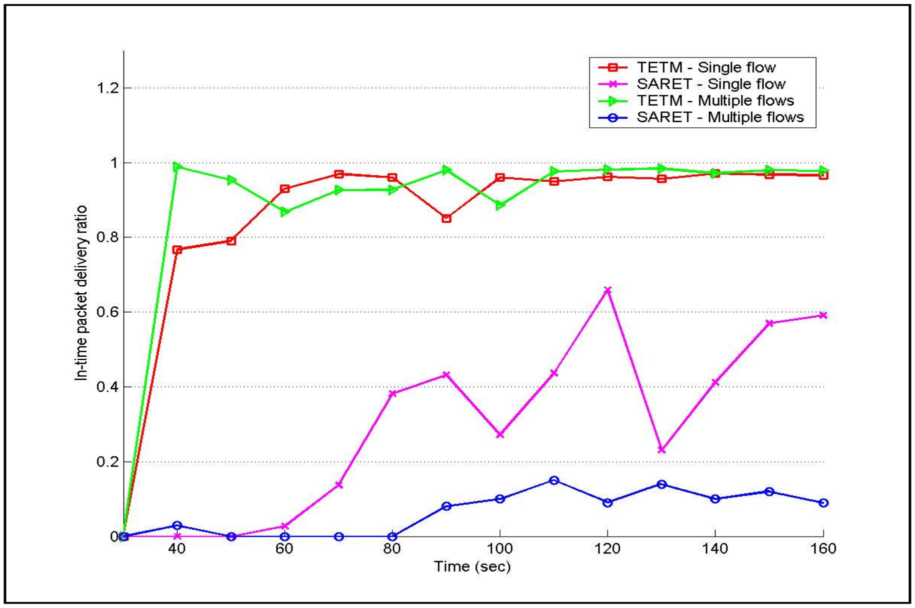

Figure 7.23 In-time packet delivery ratios of 100 event nodes randomly deployed in a single flow and multiple flows reporting an event using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

For single flow one first hop node routes event information to actor and for multiple flows three nodes are routing event information. In both these scenarios, the packet delivery ratio of TETM is considerably high (above 90 %), this is because the schedule based packet forwarding scheme decreases packet drops due to interference.

SARET performs poorly in case of single flow, as the sudden impulse of event information from 100 event nodes routing through a single node (first hop), results into congestion (Figure 7.23). Also, the congestion signals fail to reach all the sources resulting into further increase in the reporting rates nodes. As a result congestion persists in the network. When using SARET in multiple flows, few congestion signals can reach the destination because the load of the network is shared by three nodes. Therefore, the degree of congestion is less than in case of single flow, resulting into higher throughput (Figure 7.23).

Likewise, the per node throughput of SARET in case of multiple flows is better than in case of single flow, as shown in Figure 7.24. Since schedule based forwarding decreases interference, the throughput of TETM is higher than SARET for both single and multiple flows.

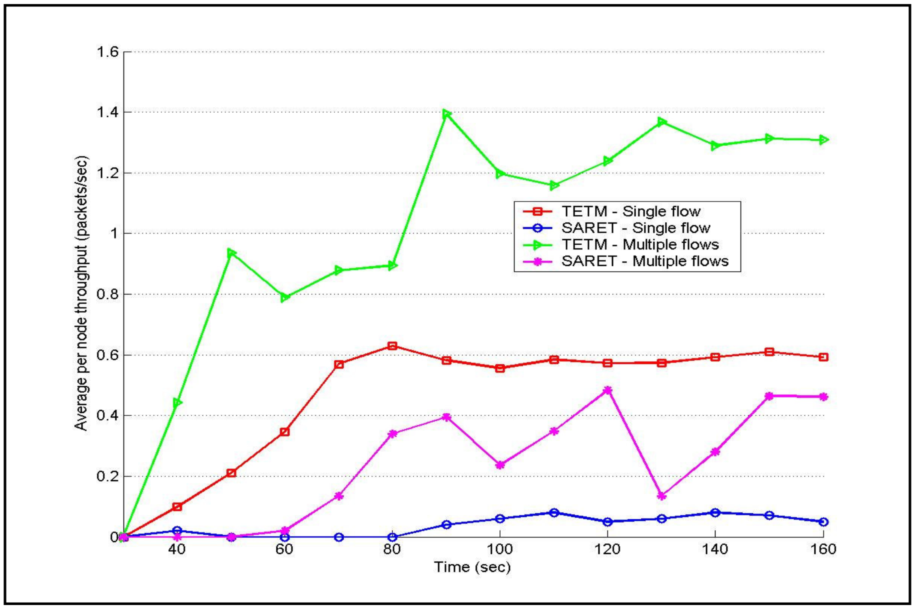

Figure 7.24 Average per node throughput of 100 event nodes randomly deployed in a single flow and multiple flows reporting an event using Time-bound Event Transport Mode (TETM) and Simple sensors-to-Actors Real-time Event Transport (SARET).

CHAPTER EIGHT
ACTOR TO SENSORS RELIABLE TRANSPORT

8.1 Overview

The information flow from a single sink to all nodes in Wireless Sensor Networks (WSNs) is named as *sink-to-sensors* or *one-to-many* information flow. Instead of disseminating the information from a single destination (sink) to all sensors, in Wireless Sensor and Actor Networks (WSANs), this information flow can be subdivided into multiple one-to-many flows, using actor nodes. Each flow from an actor to its member sensor nodes is considered as a separate information flow.

The contents of information flow depend on the nature of application in sensor networks. The actor to sensor information flow can be associated with querying the sensor nodes for updating the actor's view of the sensor field. Also, the same information flow can be used to either re-task or customize the binary codes running on the sensor nodes.

For example, in the case of mining application, re-tasking may be needed to reprogram certain groups of sensors (e.g., within a disaster recovery area). Re-tasking allows sensor nodes which were deployed for monitoring environmental conditions to perform a different sensing task. This would require addressing groups of sensors, loading new binaries into them, and then, switching over to the new re-tasked application in a controlled manner. Another example of reliable one-to-many information flow, relates to simply injecting scripts into sensors to customize them rather than sending complete, and potentially bandwidth demanding, code segments. Re-tasking becomes increasingly challenging with the increase in the numbers of sensor nodes in the network.

The challenges associated with applications, such as re-tasking, include the transport of information to possibly hundreds or thousands of nodes in a controlled, reliable, robust and scalable manner. Since the sensor nodes have limited energy, the transport mechanism needs to be energy-efficient. The error rates experienced in sensor networks can vary widely, and therefore, any reliable transport protocol must be capable of delivering reliable data to large numbers of sensor nodes under such conditions.

In the existing literature, one of the key one-to-many transport protocols for WSNs is Pump Slowly Fetch Quickly (PSFQ) (Wan, Campbell, & Krishnamurthy, 2005) protocol. The basic idea of PSFQ is to distribute data from a source node by pacing data at a relatively slow speed (pump slowly), but allowing nodes that experience data loss to fetch (i.e., recover) any missing segments from immediate neighbors very aggressively (local recovery, fetch quickly).

In this study, an Actor to Sensor Reliable Transport (ASRT) protocol is presented. ASRT is aimed to provide guaranteed information transport with minimum energy expenditure. The basic design of ASRT protocol complements the design of PSFQ protocol. Like PSFQ, ASRT uses in-sequence data forwarding and a NACK (Negative ACKnowledgement) based data recovery mechanism. However, the operation of ASRT is different from PSFQ and this study will try to prove that the pump slowly and fetch quickly operation of PSFQ is only suitable for sparse networks with low density. This study also suggests that the rate at which errors (missing packets) are recovered (NACKs sent) depends on arrangement of nodes in the network.

8.2 Protocol Design and Operation

The design of the ASRT protocol is based on the definition of reliability for actor to sensors transport. Reliability in ASRT protocol is defined as *the complete transfer of information from a single actor to all member sensor nodes; under variable channel error rates*. Information that is to be transported in ASRT protocol is a binary file. It is divided into segments and each segment contains packets equal to the buffer size of the sensor nodes. Furthermore, a unique sequence number in incremental order is assigned to each packet of a segment by the actor node.

Since actor to sensors information flow is in the opposite direction from sensors to actor flow, the definitions for next hop and previous hop nodes are redefined. In Figure 8.1, the direction of information flow in actor to sensors transport is shown.

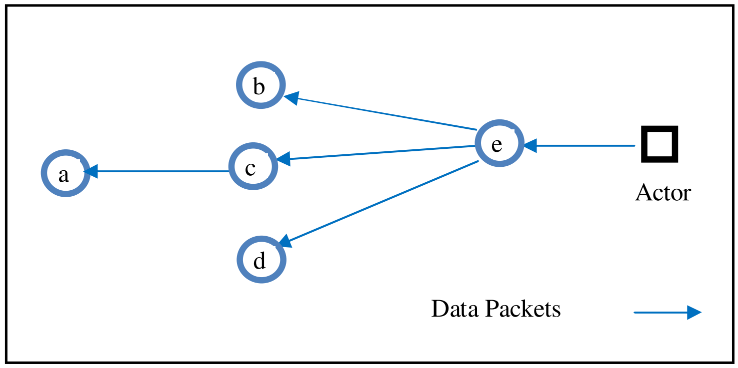

Figure 8.1 Direction of information flow in actor to sensors transport.

Node *b, c,* and *d* are the *next hop* or *child nodes* of node *e* while *e* is the *previous hop* or *parent node* of nodes *b, c* and *d*. The ASRT protocol takes advantage of the existing framework proposed in the sensors to actors transport. Using the actor selection procedure, as explained in section 4.6, nodes obtain membership from an actor and determine the hop distance from the actor. Likewise, the proposed transport solution provides both next hop and previous hop tables. The design of SARET protocol is comprised of data dissemination and error recovery mechanisms.

8.2.1 Data Dissemination

Data can be forwarded (disseminated) to all the sensor nodes using a flooding scheme. Each node receiving a new packet can broadcast the packet to its neighboring nodes. In order to prevent infinite loops, certain TTL factor can be assigned to packets. The disadvantage of a flooding scheme in a scalable and dense sensor network is the extra numbers of unnecessary transmissions required to achieve coverage (Al-Karaki, & Kamal, 2004; Zhao, & Govindan, 2003).

The coverage increases by rebroadcasting the same packet more than once, by the same node. However, according to the findings of Wan, Campbell, & Krishnamurthy, (2005), if a node broadcasts a packet more than four times, then the increase in the coverage from the retransmissions decreases below 4%. In wireless sensor networks, due to the high density (ranging from 20 - 60), the nodes on the same hop are close to each other. The transmissions from these nodes can be considered to be from same node. In this situation, it is not energy efficient that upon the reception of a new packet all nodes broadcast the same packet, as the coverage does not increase considerably.

A modified flooding scheme is used in ASRT protocol for data dissemination. The aim of this scheme is to transport the data in the network using minimum number of transmissions and achieving maximum coverage. A Data Dissemination Timer (DDT) is used by the nodes to forward a packet. Data dissemination of ASRT protocol is described below:

- Upon the reception of a new in-sequenced packet, nodes randomly select a value between 0 and DDT_{max}, where DDT_{max} is the maximum limit of DDT. At the expiry of this timer, nodes broadcast the newly received sequenced packet for their neighboring nodes.
- Before the expiry of the DDT timer, if a node hears four broadcasts of the same packet from the neighboring nodes, then it cancels its broadcast. This allows achieving coverage while decreasing the number of retransmissions.
- More than one node can broadcast information to node *a* in Figure 8.1, at the expiry of DDT timer. However, in ASRT nodes which do not have any next hop node (e.g., nodes *b* and *d*) are named *end nodes*, as they do not take part in data forwarding. This decreases the number of broadcasts and helps to terminate a packet from the network.

In wireless environments, nodes can broadcast newly received packets using either in-sequence forwarding or out-of-sequence forwarding. In the proposed data dissemination scheme, an in-sequence packet forwarding policy is used. In case of in-sequence forwarding, if a packet with sequence number greater than the previously received packet is received with no sequence gap, then the node will forward this packet to the neighbor nodes. Otherwise, if an out-of-sequence packet is received, then the node will not forward this packet and will immediately trigger the packet recovery procedure. Multi-hop in-sequence packet forwarding is shown in Figure 8.2, in which node 2 does not forward out-of-sequence packet 3, until packet 2 is received.

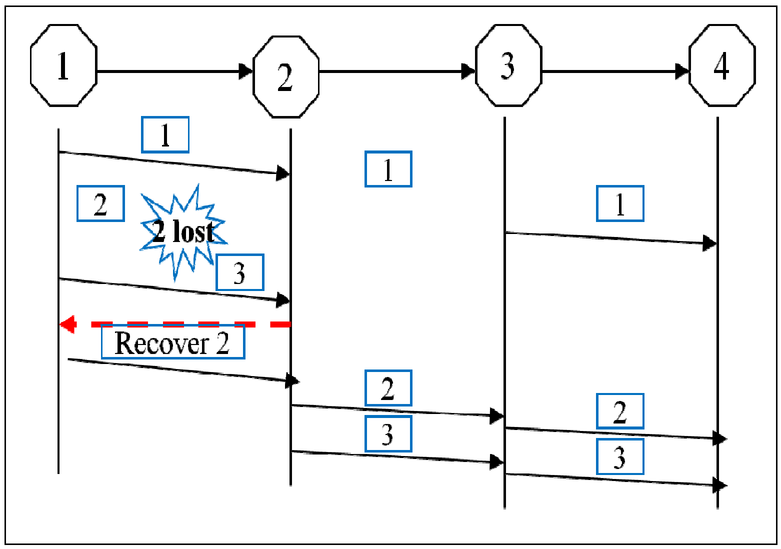

Figure 8.2 Multi-hop in-sequence packet forwarding.

For out-of-sequence packet forwarding, nodes can forward both in-sequence and out of sequenced received packets to their neighbor nodes. This allows quick dissemination of packets in the network but will result in extra transmissions and energy consumption, as each node receiving a sequence gap will forward the NACK packets. This is shown in Figure 8.3, where node 2 forwards out-of-sequence packet 3 to next hop nodes, resulting into the broadcast of additional transmissions (NACKs) from node 3 and 4.

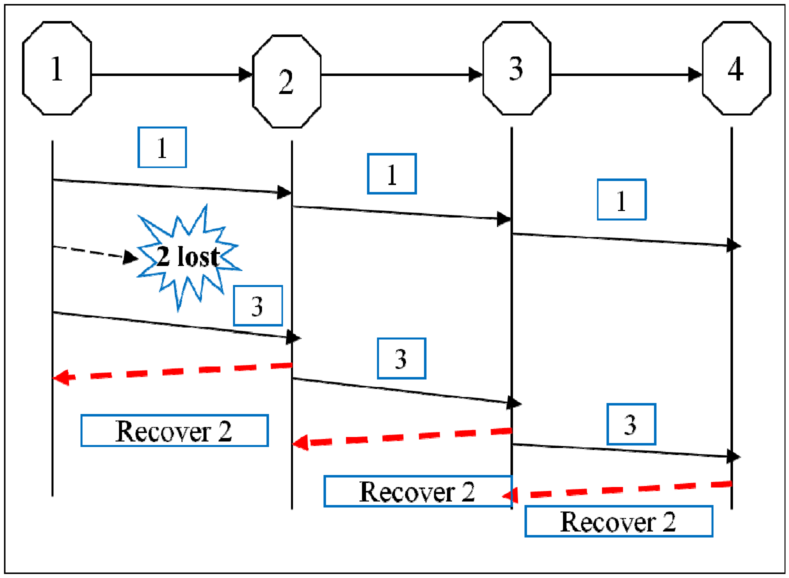

Figure 8.3 Multi-hop out-of-sequence packet forwarding.

8.2.2 Error Recovery

The poor quality of wireless links in sensor networks results into packet drops, which are variable depending on the link quality. Since packets in ASRT contain sequence numbers, a missing sequence number indicates a packet drop.

In order to recover dropped packets, either an end-to-end or hop-by-hop error recovery model can be used. According to the findings of Wan, Campbell, & Krishnamurthy, (2005), the former is not appropriate for the multi-hop error prone communication in wireless sensor networks. Since the destination can be at a multiple hop distance from the source, successful delivery of a packet to a destination is difficult. Using an end-to-end error recovery model in scalable sensor networks will result into NACK implosion, as the NACKs from hundreds of sensor nodes will be sent to a single source. On the other hand, hop-by-hop packet delivery divides the multi-hop packet transport into a number of single hops. Each previous hop acts as the source and the next hop as the destination. Thus, a packet drop at any hop can be immediately detected and recovered. Therefore, in ASRT protocol a hop-by-hop error recovery model is used.

If a sequence number gap is observed by nodes upon the reception of a new packet, nodes will broadcast a NACK packet. The neighboring nodes upon the reception of the NACK packet will broadcast the required packet; these packets are named as *NACK response packets*. A node in ASRT caches all segment packets in the buffer, until it receives a *segment receipt packet* from all of its neighboring nodes. By issuing a segment receipt packet, a node informs the successful reception of all segment packets. Since nodes in ASRT caches all the transmitted packets of a segment, NACK packets are not further broadcasted by nodes. This helps to remove NACK implosion problem. The format of a NACK packet used in ASRT protocol is shown in Table 8.1.

Table 8.1 NACK packet format in actor to sensor reliable transport protocol.

2 Byte	2 Bytes	2 Bytes	2 bits	2 bytes	2 Bytes	2 Bytes	2 Bytes
Source ID	Destination (Broadcast)	TTL (1)	PKT_TYPE (NACK)	Segment ID	Start sequence number	End sequence number	Header length

The fields of the NACK packet are explained below:

- Source ID: The identification number of the node broadcasting a data packet.
- Destination: Broadcast address (-1).
- Time to Live (TTL): A node broadcasts a packet only for its neighboring nodes. Therefore, the value of TTL is set to 1.
- PKT_TYPE: This field indicates the type of the packet, which is a NACK packet.
- Segment ID: The segment ID indicates that the missing packet(s) belong to this segment number.
- Start sequence number: The sequence number of a missing packet or the start sequence number of consecutive missing packets.
- End sequence number: The last sequence number of consecutive missing packets. In case of a single missing packet, both start and end sequence number fields include same sequence number.
- Header length: The length of the NACK header.

Since a node receiving a NACK does not broadcast it further, the TTL factor in the NACK packet is 1. If a node has consecutive missing packets then it can send a single NACK to indicate all the missing packets. For example, if a node receives in-sequence packets from 1 to 10 but later it receives packet 14, then it will send NACK with start sequence number 11 and end sequence number 13. In NACK response, a node having all the missing packets will broadcast the requested packets in a single go, as it receives the NACK packet. A number of nodes can hear the NACK and depending on their in-sequenced packets can broadcast in response to the NACK. In order to limit the number of broadcasts, nodes use a random timer value much less than NACKT and at the expiry of this timer broadcast the NACK response packets. Neighboring nodes overhearing this response before their NACK response transmission, will cancel their transmissions.

A NACK Timer (NACKT) is used to broadcast NACKs in ASRT protocol. Nodes on the detection of a missing packet set their NACKT and upon the expiry of this timer, nodes will broadcast their NACK packets. The possibility that more than one node does not receive a broadcasted packet depends on channel conditions. In dense networks, the numbers of nodes within the radio range of a single node can range from 40 to 60. In this case nodes receiving an out-of-sequence packet will broadcast same NACK packet.

ASRT reduces the number of same NACK transmissions from neighboring nodes by inserting small random delays between NACK broadcasts. As a missing sequence gap is detected by a node, it selects a random value of NACKT between 0 and $NACKT_{max}$. If nodes hear same NACK transmission by a neighboring node before their NACKT expiry, they will cancel their NACK transmission.

The length of NACK timer defines the quickness of the recovery process. Smaller the length of NACK timer, quicker will be the packet recovery. Longer the length of NACK timer, slower will be the recovery process. PSFQ proposed that the NACKT should be five times shorter than the DDT, aiming to quickly fetch the dropped packets. In dense networks with the increase in the number of nodes, the nodes forwarding data packets will also increase.

Since a packet in dense arrangement can be broadcasted several times, broadcasting NACKs quickly will increase the load on the network. Also, this will result in extra transmissions and increased energy consumption. It will be shown in the simulations of ASRT protocol that for dense and scalable node arrangements NACKs should be sent at a slower rate while in sparse node arrangements NACKs should be sent quickly. Since the value of NACKT is dependent on node arrangement, in ASRT protocol the length of NACKT is broadcasted with the first data packet of the segment.

The transmission of a NACK packet does not guarantee that the node will receive the missing packet. Since the NACK response packet and NACK packet itself can be dropped, due to channel errors. Therefore in ASRT protocol, if a node does not receive a NACK response after broadcasting a NACK packet, it can transmit the same NACK three times.

A natural drawback of NACK based schemes is that a NACK packet can only be transmitted on the reception of an out-of-sequence packet. If a node initially receives a few in-sequenced packets but later no packet is received, then in pure NACK based scheme it is not possible to detect the packet loss. Therefore, in ASRT protocol like PSFQ an Advance NACK Timer (ANACKT) is used to detect such packet losses. When a node receives the first sequenced packet i.e., the first packet of the segment, it sets the ANACKT. If all the packets are received in-sequence but end of file (last packet of segment) is not received then on the expiry of ANACKT, a node will broadcast NACK packets.

The value of ANACKT is based on hop distance from the actor and time required for forwarding a segment from an actor to the destination node and is calculated as:

$$\text{ANACKT} = (\text{DDT}_{max} \text{ X hop distance}) \text{ X segment length} + \mu \tag{9}$$

where μ is constant having a small value. The value of ANACKT without μ in Equation 9 represents the time required for successful delivery of a segment to the destination; in ideal conditions (no channel error). The number of times a packet is broadcasted for successful delivery is dependent on channel error conditions. Therefore, μ is used to increase the value of ANACKT. For precise calculation of ANACKT, the value of μ should be mirrored according to average channel error rate observed in the network.

8.2.3 Status of Transport

In the transport of a binary file from the source (actor/sink) to the destinations (sensors), it is required that the source must be updated with the status of transport. The use of an acknowledge (ACK) per single data packet will result in ACK implosion and is also expensive in terms of energy utilization. Therefore, a simple ACK based scheme for updating the transport status of a source is not suitable in sensor networks.

Two methods for updating the transport status of a source are used in ASRT protocol; segment receipt packets and explicit transport status monitoring. These methods are explained below:

- *Segment receipt packets:* These packets are an implicit method for updating the transport status of network nodes and the source node. When a node receives all the data packets of segment, then it broadcast the segment receipt packet to its neighbor nodes. The neighboring nodes in their neighbor node table updates the status of the node transport for the specific node as completed. Moreover, the parent node will also send this information to the actor node. In this manner, the source will be updated by all the nodes in the network about their status of transport.
- *Explicit transport status monitoring:* Due to high channel error rates, it is possible that nodes do not receive all the data packets or they receive the packets after a long delay. In this condition, the source can explicitly

broadcast a *transport status request packet* to all the nodes. This packet is broadcasted in the network to the end nodes (destination). The end nodes upon the reception of this packet will not further broadcast the packet. Moreover, the end nodes will initiate the reply for this packet, by broadcasting their node identification number and last sequenced number received in the *status reply packet*. All the intermediate nodes receiving status reply packet will piggy back their node identification number and last sequenced number in this packet, and will further broadcast the packet.

8.3 Simulation Results

The performance of the proposed Actor to Sensors Reliable Transport (ASRT) is observed using network simulator NS-2. The simulation scenario is comprised of a wireless sensor network, with different numbers of sensor nodes deployed in a 100 x 100 m field. The basic simulation parameters are shown in Table 8.2.

Table 8.2 Simulation parameters.

Transport Layer	ASRT
Network Layer	Broadcast
MAC Layer	802.11
Propagation model	Two-ray reflection
Deployment	Random
IFQ Length	65 Packets
Transmit Power	0.660 W (fixed)
Receive Power	0.395 W
Radio Range	20m
1 Segment	50 Data packets
DDT_{max}	0.02 seconds
NACKT	variable

A single segment is broadcasted by an actor into the network and the segment length is 50 packets. An in-sequence data packet is broadcasted by the actor in the network after every 0.01 seconds, until all of the 50 packets are transmitted. The transport starts at 20 seconds into the simulation time and ends at 100 seconds.

The performance of ASRT is evaluated at different channel error rates, in terms of reliability, total number of transmissions required for successful transport, latency and energy consumption. An important factor in the design of ASRT protocol is the length of NACK interval. Since NACKs are a natural overhead in the design of ASRT protocol, the length of NACK interval is calculated in order to decrease the total number of transmissions. The performance of ASRT using different NACK interval lengths and under different channel error rates is observed. Also, the length of NACK intervals considered in the simulation results are relative to the length of data dissemination timer.

The efficiency of ASRT is compared with Pump Slowly Fetch Quickly (PSFQ) protocol (Wan, Campbell, & Krishnamurthy, 2005). In PSFQ, the fetch operation which is triggered on the detection of a missing sequence number is five times faster than the pump operation. Therefore for PSFQ, the value of NACKT is 0.004 seconds. Different node arrangements (scenarios) are used to precisely evaluate the performance of both ASRT and PSFQ, which are given below:

- *Linear node arrangement:* In this scenario, five nodes are linearly arranged in separate hops. The actor is located at coordinates (90,50) and the sensor nodes 1, 2, 3, 4 and 5 are at coordinates (81,50),(72,50),(63,50),(54,50) and (45,50) respectively. The radio range used in this simulation is 10 meters.
- *Sparse node arrangement:* This node arrangement is similar to the one shown in Figure 6.6.
- *Dense node arrangement:* In this scenario, 50 nodes are arranged at the same hop in such a way that the node density is 50. The actor is located at coordinates (90,50) and all the sensor field is centered at coordinated (80,50) with a diameter of 20 meters.
- *Scalable node arrangement:* In this node arrangement, 100 sensor nodes are randomly distributed within a sensor field of 100x100 meters, with actor at (90,50) coordinates.

8.3.1 Linear Node Arrangement

The time required for the successful transport of all data packets under uniform channel error rates of 10%, 30%, 50% and 70% in linear node arrangement are shown in Figures 8.4, 8.5, 8.6 and 8.7 respectively.

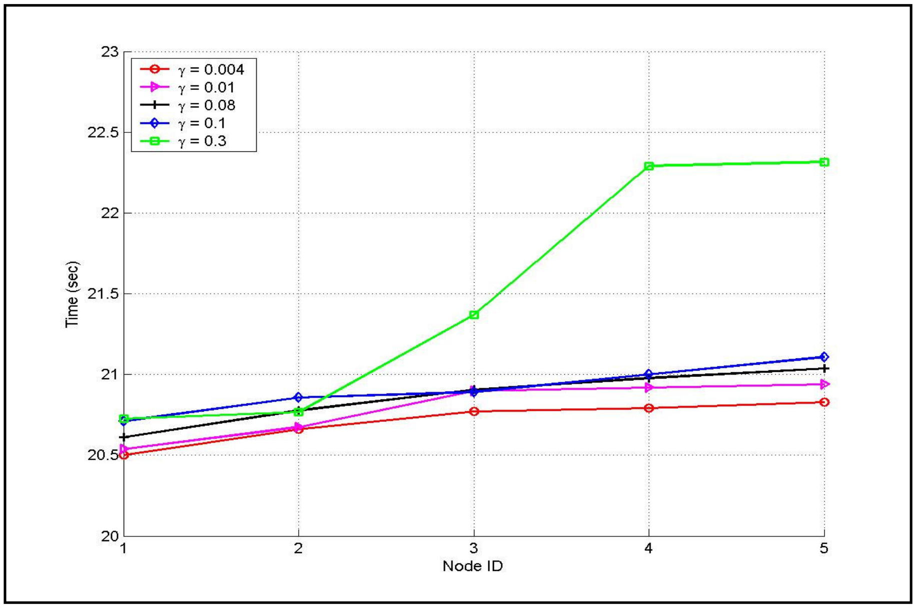

Figure 8.4 Per node latency of transport under channel error rate of 10% for 5 nodes linearly arranged, using different NACK interval lengths (γ).

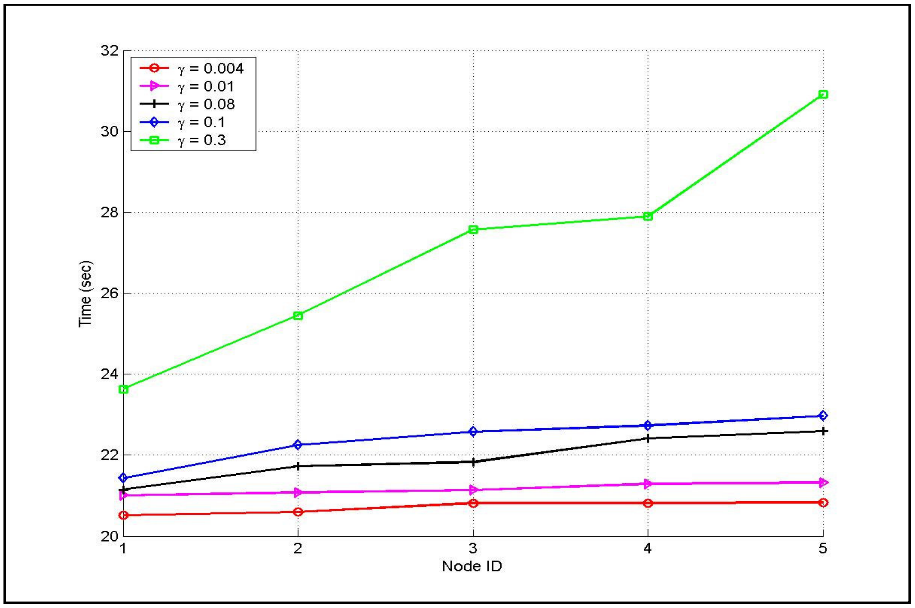

Figure 8.5 Per node latency of transport under channel error rate of 30% for 5 nodes linearly arranged, using different NACK interval lengths (γ).

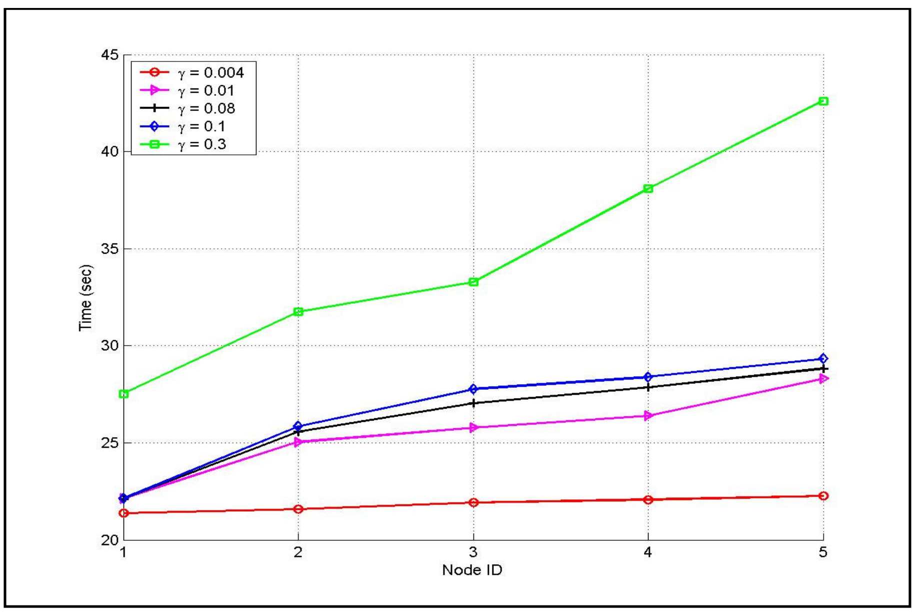

Figure 8.6 Per node latency of transport under channel error rate of 50% for 5 nodes linearly arranged, using different NACK interval lengths (γ).

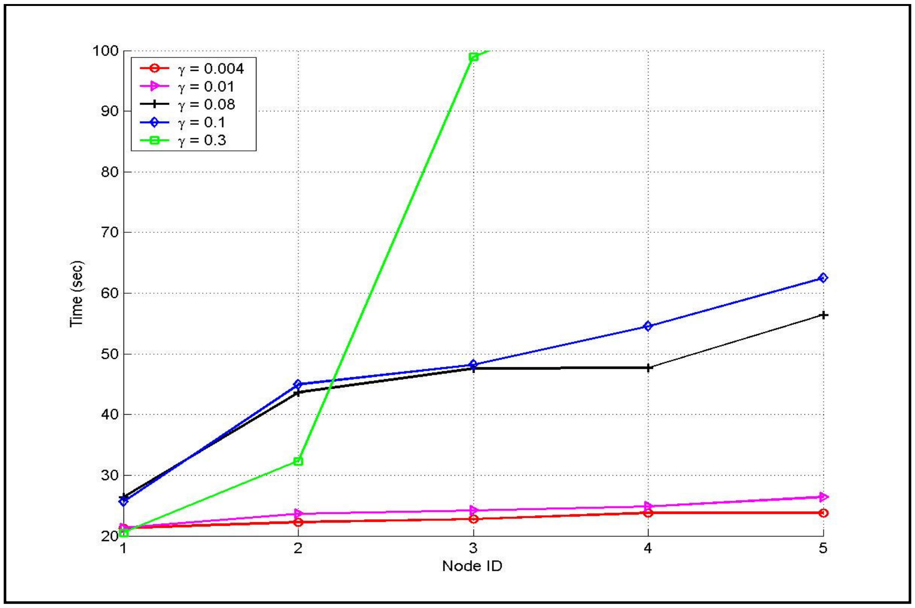

Figure 8.7 Per node latency of transport under channel error rate of 70% for 5 nodes linearly arranged, using different NACK interval lengths (γ).

In the linear node arrangement, a node can only receive new and missing data packets from only one previous hop node. As a result, by decreasing the length of NACK interval decreases the latency of transport (Figure 8.4, 8.5, 8.6 and 8.7). Lowest latency is achieved when the NACK timer ($\gamma = 0.004$) is five times smaller than the dissemination timer (0.01), similar to the findings of PSFQ (Wan, Campbell, & Krishnamurthy, 2005). However, decreasing the length of NACK interval will increase the total number of transmissions and energy consumption of nodes. This is evident from Figures 8.8 and 8.9 in which the total number of transmission required for successful transport of data packets and the energy consumed are shown, respectively.

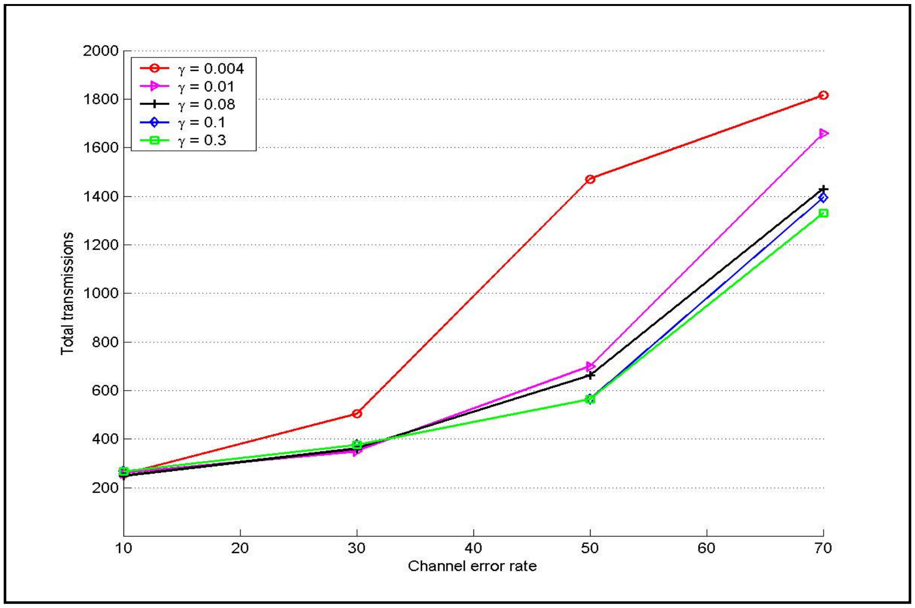

Figure 8.8 total numbers of transmissions required for successful transport of data packets to 5 nodes under different channel error rates and NACK interval lengths (γ).

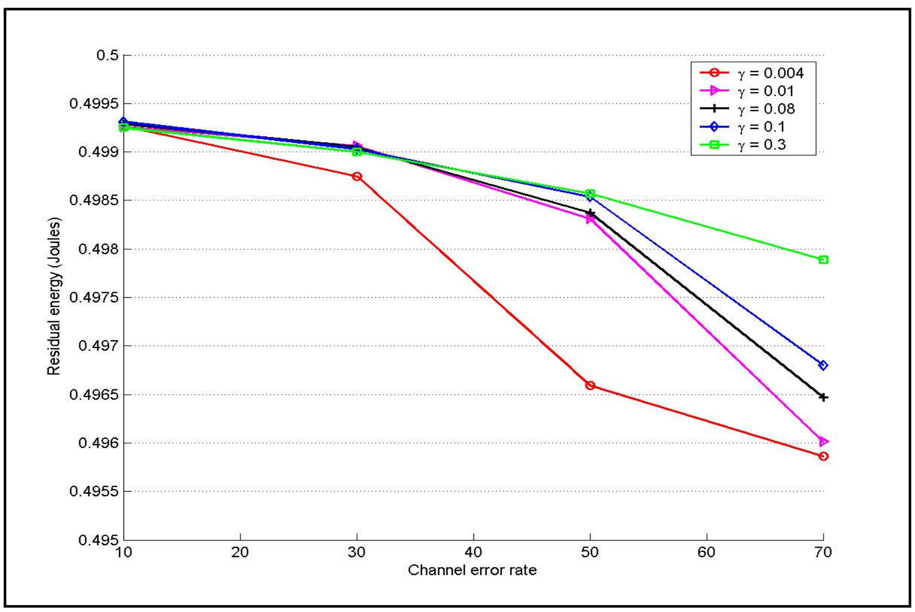

Figure 8.9 Residual energy of network after successful transport of all the data packets to 5 nodes under different channel error rates and NACK interval lengths (γ).

8.3.2 Sparse Node Arrangement

The time required for the successful transport of all data packets under uniform channel error rates of 10%, 30%, and 50% in sparse node arrangement are shown in Figures 8.10, 8.11 and 8.12 respectively.

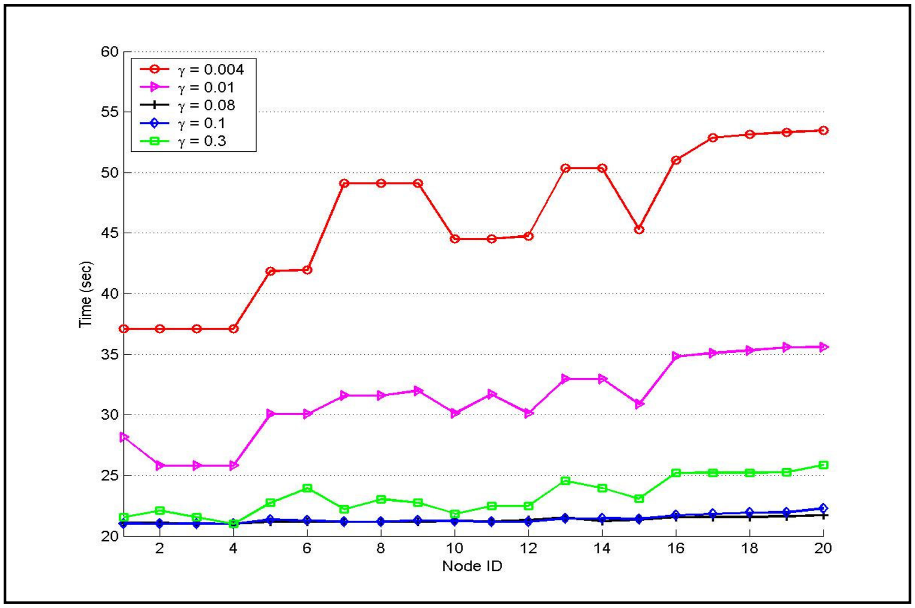

Figure 8.10 Per node latency of transport under channel error rate of 10% for 20 nodes sparsely arranged, using different NACK interval lengths (γ).

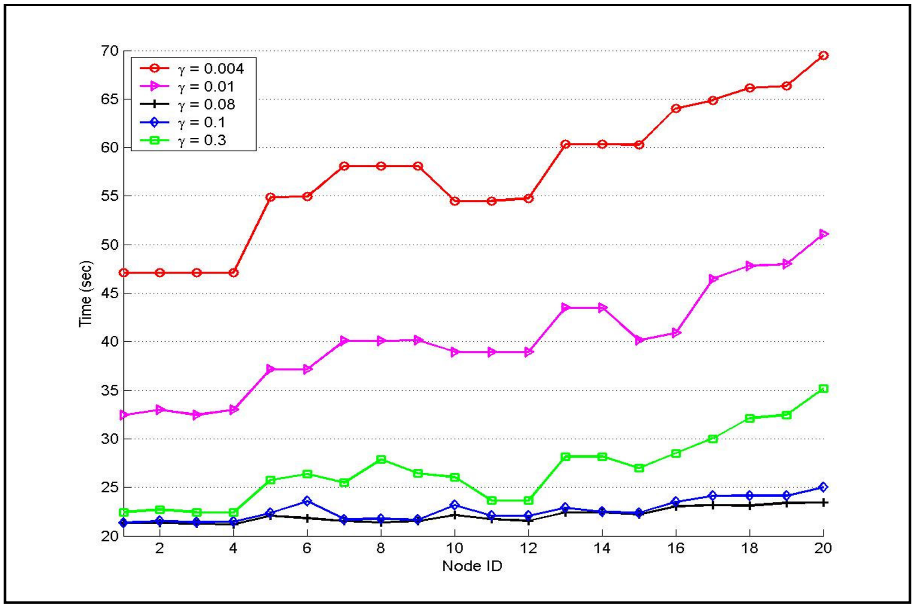

Figure 8.11 Per node latency of transport under channel error rate of 30% for 20 nodes sparsely arranged, using different NACK interval lengths (γ).

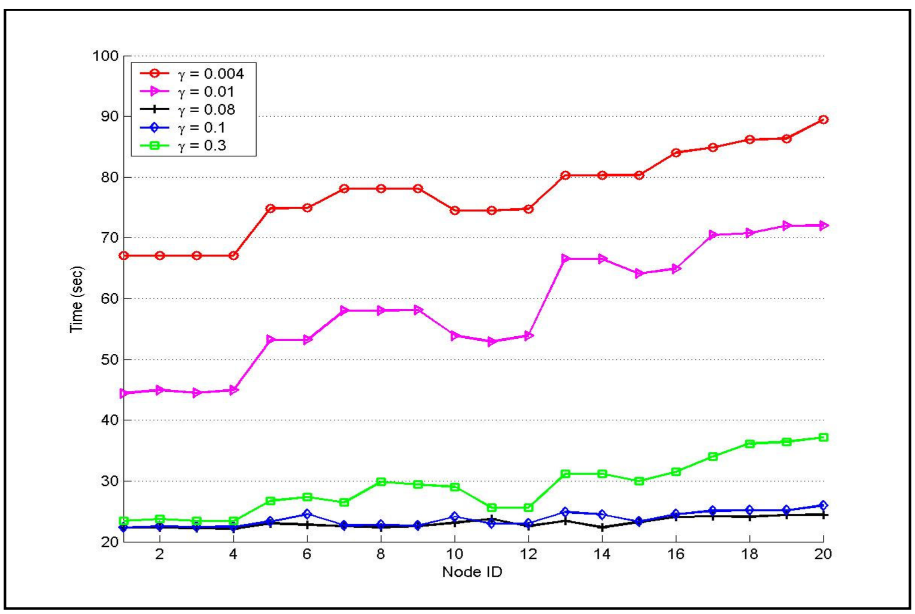

Figure 8.12 Per node latency of transport under channel error rate of 50% for 20 nodes sparsely arranged, using different NACK interval lengths (γ).

Latency of transport is affected by the load on the network. Smaller the length of NACK interval greater will be the number of transmissions and higher will be the load on the network. As a result, the latency of transport is higher in case of γ =

0.004 at all channel error rates (Figure 8.10, 8.11 and 8.12). According to the findings of this study, in a multi-hop sparse node arrangement, latency of transport is best when the NACK interval length is four (0.08) and five (0.1) times greater than the dissemination interval length (0.02).

Since a node can hear the transmission of more than one neighboring nodes, same packet can be received from multiple nodes. As a result, delaying the NACK transmission not only decreases latency but also allows decreases the number of transmissions and energy consumption of the nodes in the network. The total number of transmissions required for successful transport of data packets and the consumed energy are shown in Figures 8.13 and 8.14.

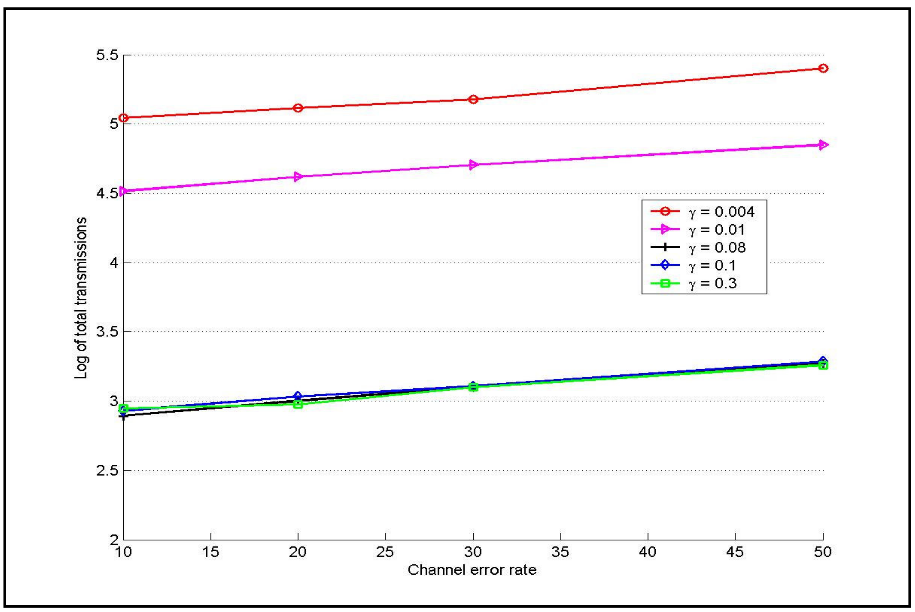

Figure 8.13 Log of total number of transmissions required for successful transport of data packets to 20 nodes under different channel error rates and NACK interval lengths (γ).

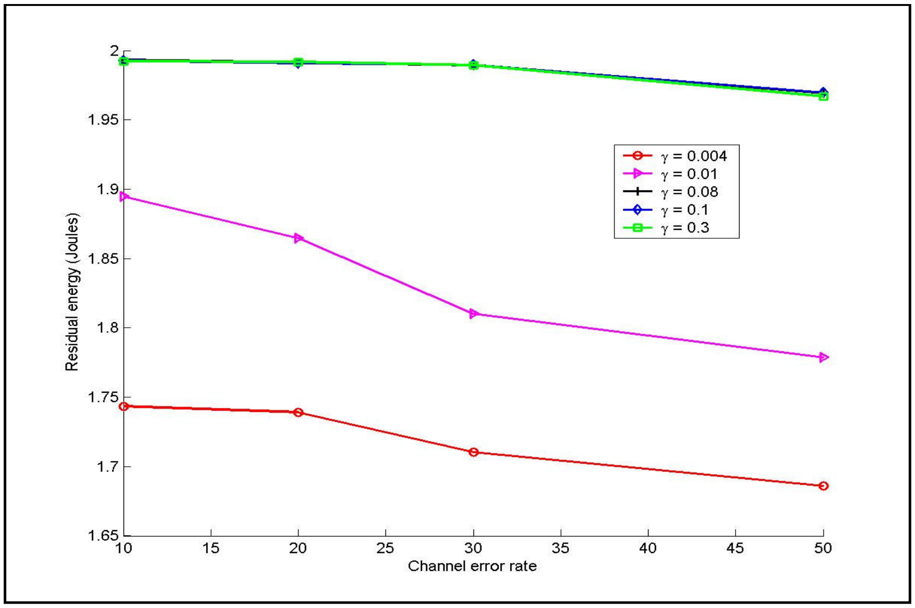

Figure 8.14 Residual energy of network after successful transport of data to 20 nodes under different channel error rates and NACK interval lengths (γ).

In the sparse node arrangement, the performance of ASRT is compared with PSFQ in terms of fixed NACK transmissions. PSFQ uses a NACKT of 0.004 second, while ASRT uses a NACKT of 0.08 second. Three different results are shown in Figure 8.15, for per node allowed total NACKs of 1, 3 and 5 at channel error rates of 10%, 30% and 50%.

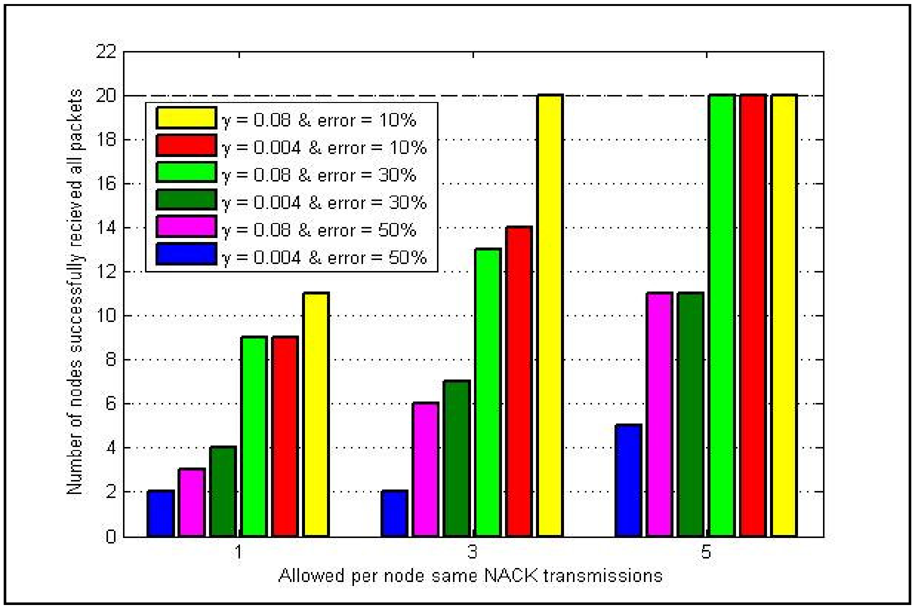

Figure 8.15 **S**uccessful transports of data to 20 nodes using constant number of NACK transmissions and at different channel error rates.

Since data dissemination of all nodes is not synchronized, each node broadcast sequenced packets at different intervals. As a result, when slow NACKT is used the probability that nodes receive packets from neighboring nodes without sending NACKs increases. Therefore, the successful packet delivery at NACKT 0.08 is higher than at 0.004 (Figure 8.15).

8.3.3 Dense Node Arrangement

The time required for the successful transport of all data packets under uniform channel error rates of 10%, 30%, and 50% in dense node arrangement are shown in Figures 8.16, 8.17 and 8.18 respectively.

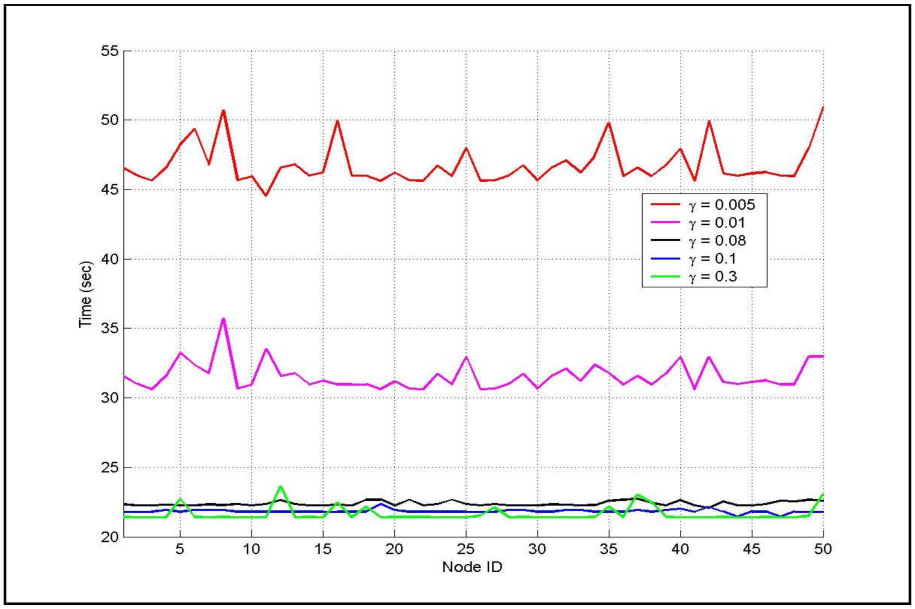

Figure 8.16 Per node latency of transport under channel error rate of 10% for 50 nodes densely deployed, using different NACK interval lengths (γ).

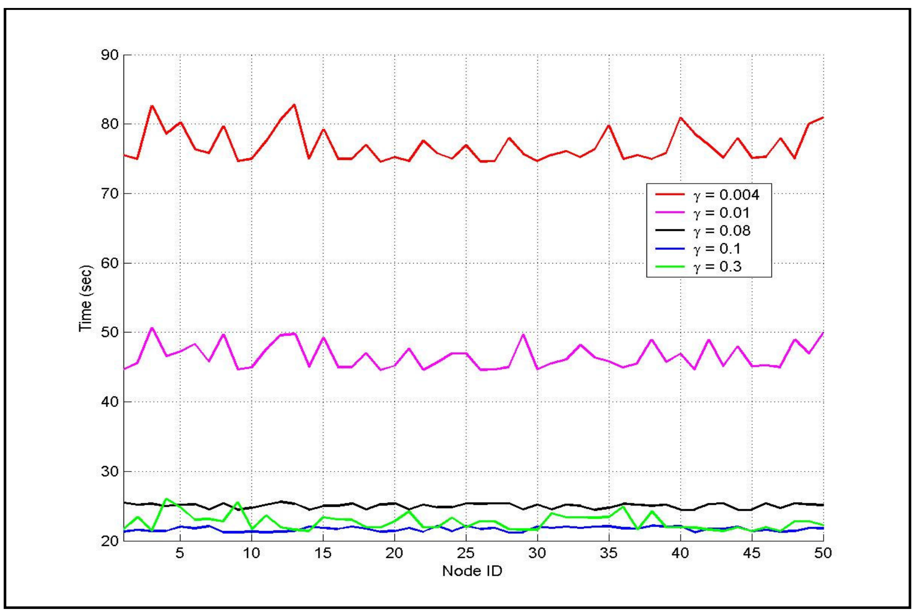

Figure 8.17 Per node latency of transport under channel error rate of 30% for 50 nodes densely deployed, using different NACK interval lengths (γ).

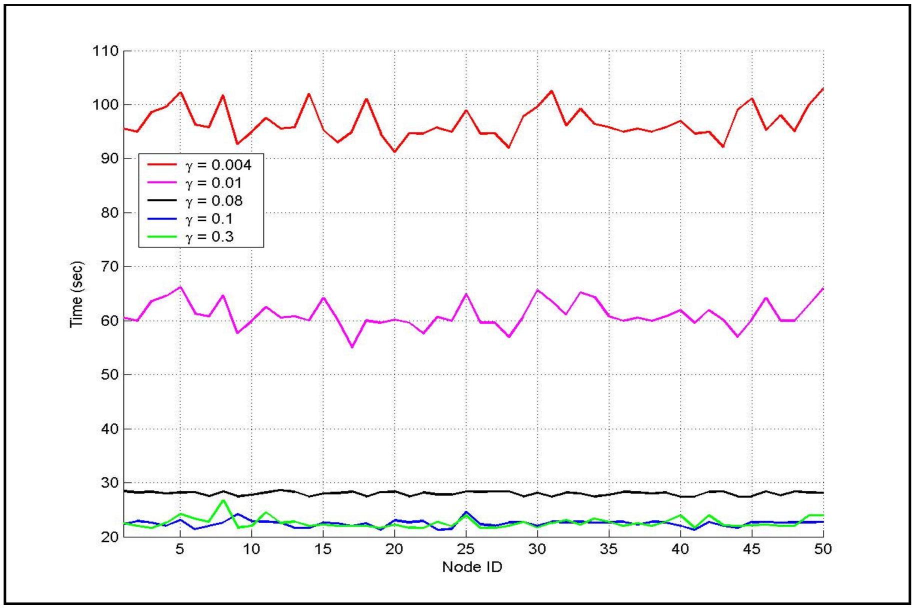

Figure 8.18 Per node latency of transport under channel error rate of 50% for 50 nodes densely deployed, using different NACK interval lengths (γ).

Since the number of same packet transmissions increase in high density scenario, the latency of transport is less for high NACK interval lengths (Figures 8.16, 8.17 and 8.18). For NACK interval length of 0.004, as considered by PSFQ, the latency increases as congestion occurs and packets are dropped at nodes. This is because in this case of a lot of NACKs and NACK response packets are generated. Since all nodes are on the same hop, transmissions are also affected by interference in dense arrangement.

The total number of transmissions required for successful transport of data packets and the energy consumed in dense node arrangement are shown in Figures 8.19 and 8.20. A short NACK interval length compare to data dissemination interval length, results in increased transmissions and energy consumption.

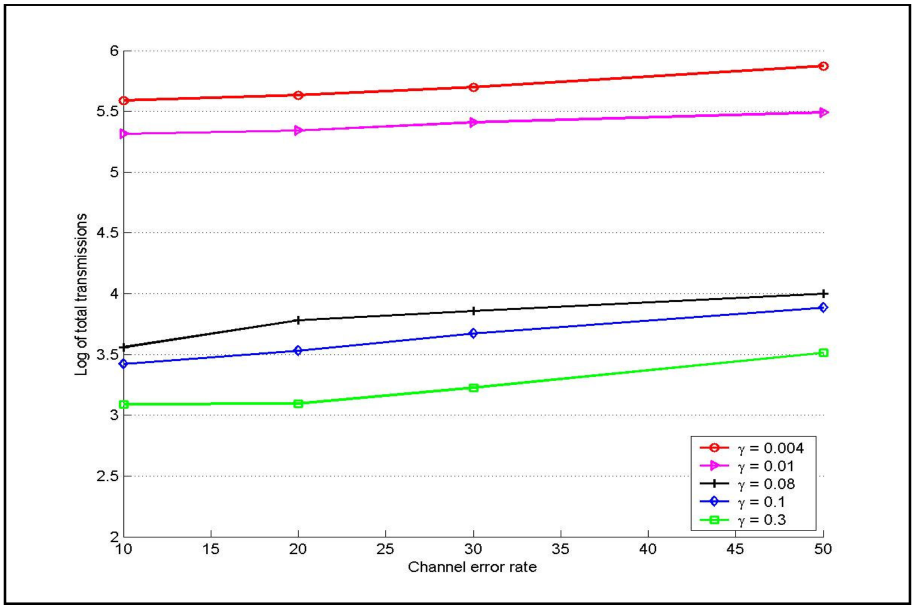

Figure 8.19 Log of total number of transmissions required for successful transport of data to 50 nodes under different channel error rates and NACK interval lengths (γ).

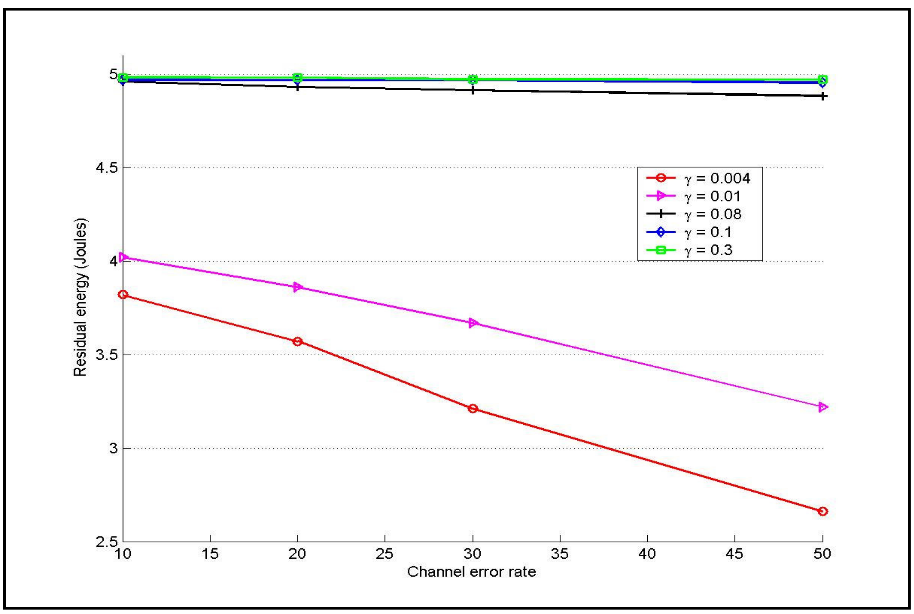

Figure 8.20 Residual energy of network after successful transport of data packets to 50 nodes under different channel error rates and NACK interval lengths (γ).

8.3.4 Scalable Node Arrangement

In scalable node arrangement, the number of nodes with the sensor field is 100. As the number of nodes increase, using short NACK interval length results in increased transmissions. Therefore during 100 second of simulation time for γ = 0.004 and γ = 0.01 seconds, the data transport is not completed. As a result, the simulations for scalable node arrangement are shown for higher values of γ. The time required for the successful transport of all data packets under uniform channel error rates of 10%, 30%, and 50% in scalable node arrangement are shown in Figures 8.21, 8.22 and 8.23 respectively.

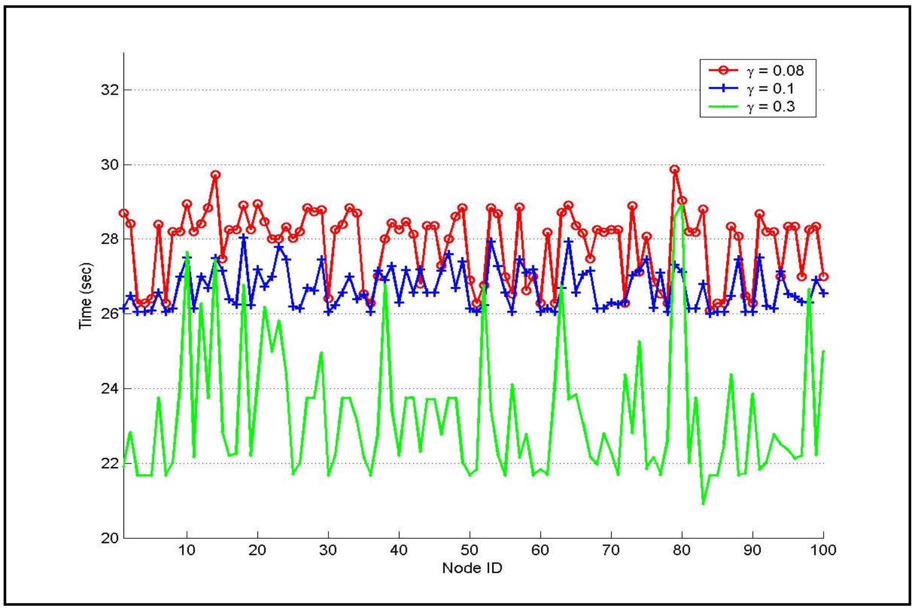

Figure 8.21 Per node latency of transport under channel error rate of 10% for 100 nodes randomly deployed, using different NACK interval lengths (γ).

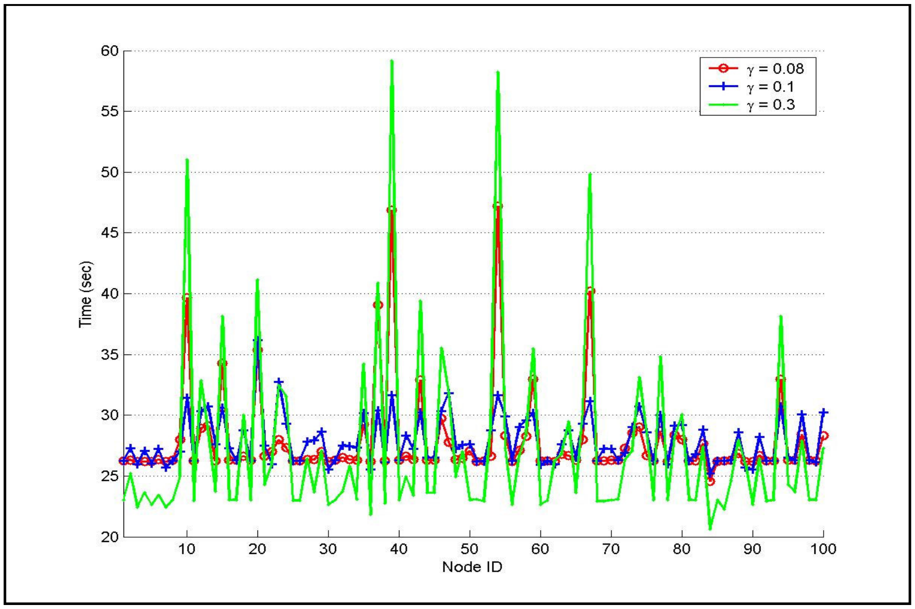

Figure 8.22 Per node latency of transport under channel error rate of 30% for 100 nodes randomly deployed, using different NACK interval lengths (γ).

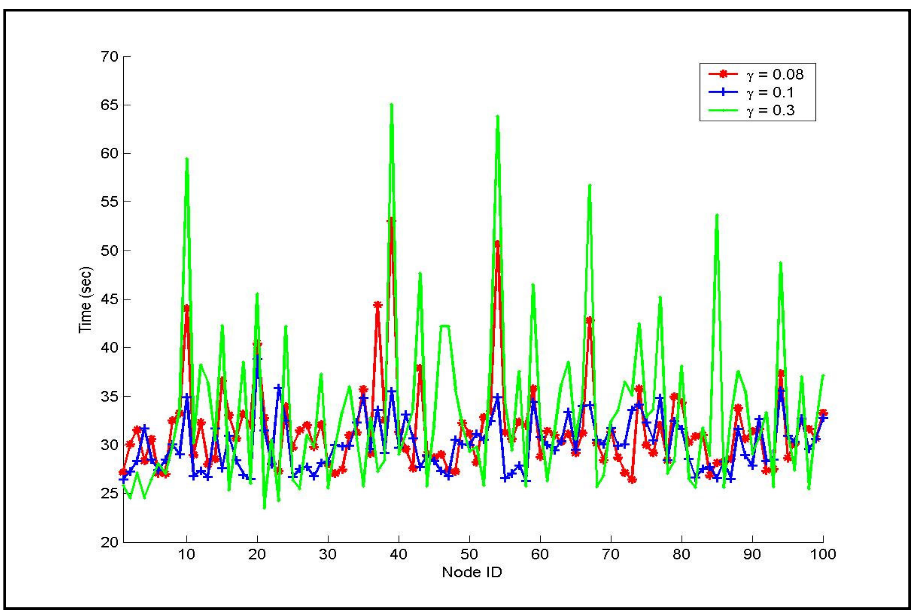

Figure 8.23 Per node latency of transport under channel error rate of 50% for 100 nodes randomly deployed, using different NACK interval lengths (γ).

In scalable node arrangement, some nodes are at multiple hop distance from the actor node and the density of nodes within the sensor field is variable. Therefore, at all channel error rates the latency observed from different nodes in the sensor field is variable (Figures 8.21, 8.22, and 8.23). Since the NACK interval lengths are pretty long as compared to the data dissemination interval, the difference in latency of transport for these NACK interval lengths is considerably less.

The total number of transmissions required for successful transport of data packets and the energy consumed are shown in Figures 8.24 and 8.25. As mentioned before, due to considerably long NACK interval lengths the difference in the performance of ASRT in these simulation results is very less.

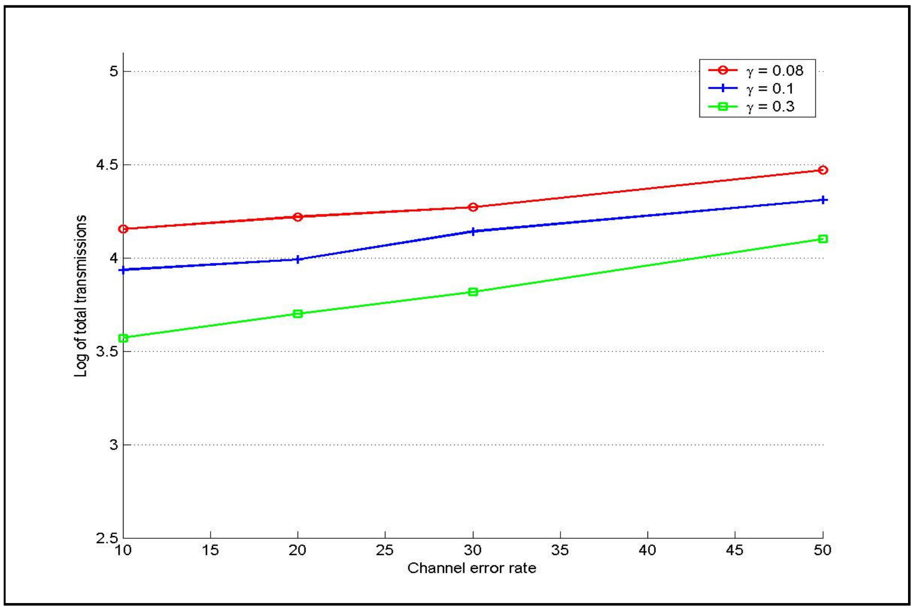

Figure 8.24 Log of total number of transmissions required for successful transport of data packets to100 nodes under different channel error rates and NACK interval lengths (γ).

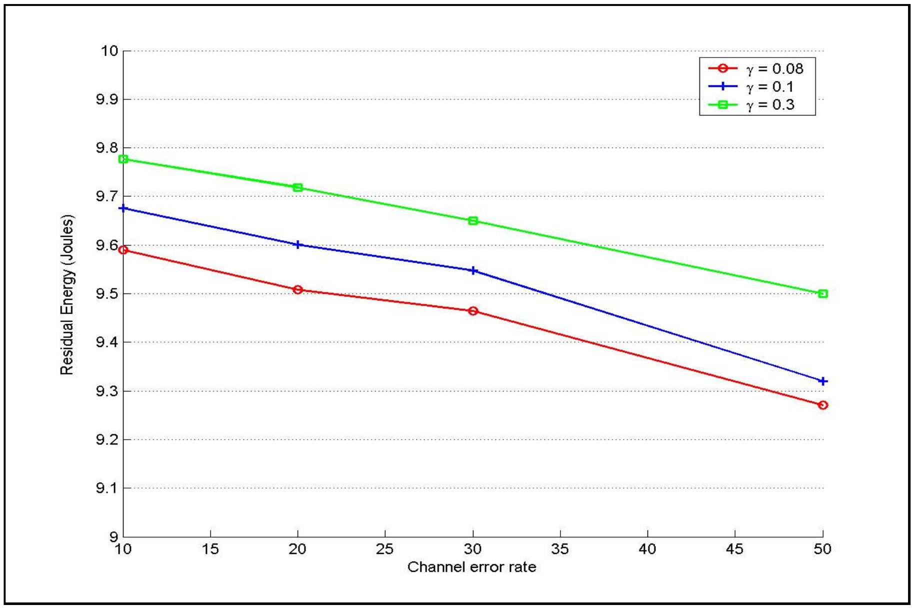

Figure 8.25 Residual energy of network after successful transport of data packets to 100 nodes under different channel error rates and NACK interval lengths (γ).

CHAPTER NINE

CONCLUSION

A reliable transport solution for Wireless Sensor and Actor Networks (WSANs) was presented in this study. The uniqueness of the proposed transport solution as compared to existing transport protocols for sensor networks is that, the proposed solution used a unified approach for the transportation of different application dependent information. The proposed transport layer for WSANs was divided into two major parts:

- Sensors-to-Actors transport
- Actor-to-Sensors transport

Sensor nodes are deployed to detect a number of application defined events. Each event has its own characteristics, like reporting rate, importance and reliability requirements. In order to achieve application defined goals for different events, the proposed sensors-to-actors transport had been divided into different transport modes. These transport modes were categorized as *simple*, *fair*, *priority* and *real-time* event transport modes. Due to the sudden impulse of data travelling from many sensor nodes to a single or few actor nodes, the event information flows in sensor networks are subject to congestion. Therefore, a congestion control scheme was presented to detect and remove congestion in the sensors-to-actors transport.

The proposed congestion control scheme was aimed to avoid and mitigate congestion while providing high throughput of the system especially in dense networks. The proposed scheme used a novel combination of hop-by-hop packet delivery time and buffer size for congestion detection and rate adjustment in sensor networks. An in-network congestion mitigation model was used in which nodes adjusted their reporting rates according to their local network congestion status. Average packet delivery time provided a realistic approach for reporting rate selection in sensor networks compared to AIMD based rate selection schemes. In order to avoid and mitigate congestion, the average packet delivery time was adjusted with buffer size of nodes, so that, nodes could provide high throughput while avoiding congestion.

In dense node arrangements, when nearby sensors suddenly start to transmit information at the same time, interference increases. Interference not only results in packet drops but also increases the congestion in the network. The proposed congestion control scheme addressed packet drops due to congestion and interference. A novel TDMA-like (schedule-based) scheme was introduced at the transport layer for orderly forwarding packets to the underlying layers. The schedule-based packet forwarding packets helped to avoid packet collisions and increased the packet delivery ratio even in high densities as compared to jittered forwarding.

Reliability in the proposed sensors-to-actors transport was measured as a ratio of total number of packets received by the destination to the total number of required packets (application defined). The proposed sensors-to-actors transport also provided maximum throughput while avoiding or controlling congestion by adjusting their reporting rates according to network conditions. The former was named as *application based reliability* while the later was named as *network based reliability*. The existing transport and event reporting protocols for sensor networks can provide either application or network based reliability. However, the proposed sensors-to-actors transport successfully provided both reliability syntaxes.

Simple Event Transport Mode (SETM) was aimed to reliably transport general event region information to the destination. Event based wireless sensor and actor networks (WSANs) are required to reliably detect an event from the sensor field and most of the applications require general event information irrespective of per event node contribution at the destination.

Fairness in wireless sensor networks demands event nodes to have an equal share in the overall throughput of the system. Fairness is difficult to achieve in sensor networks due to multiple hop packet forwarding to a single destination that results in congestion. In the Fair Event Transport Mode (FETM), sensor nodes within a single flow adjusted their reporting rates in order to fairly distribute the bandwidth among all the event reporting nodes. Fairness can be achieved by avoiding congestion and by assigning same reporting rate to all event reporting nodes. Also, a packet forwarding scheme is required to ensure fair event packet delivery from a number of sources which can be at multiple hop distance from a single destination. For these purposes, FETM used congestion control scheme based on hop-by-hop packet delivery time and buffer size of nodes. Fair reporting rate allocation was done on the basis of sub-tree size of nodes and a schedule based packet forwarding scheme was used to provide fairness.

In sensor networks depending on application needs information from a node, region or multiple events can be transported with priority. In this study, Prioritized Event Transport Mode (PETM) was designed to handle multiple events according to their application defined reporting rates. In this case, the flow bandwidth was distributed among different event reporting nodes according to their initial reporting rates. Thus, the nodes with high reporting rates obtained more priority and delivered more packets to the actor than nodes with lower rates.

In case of time critical event information, the transport solution had to ensure that the event packets must reach the destination within certain time limit, while meeting the reliability requirements. The effectiveness of an action by the actors depends on the reliability of event information in terms of the magnitude and in-time delivery of an event. For time bound transport, reliability was defined as the ratio of number of in-time packets received to the required in-time packets specified by the application within a certain time period.

Two different solutions for time bound event transport in WSANs were presented in this study: Simple sensors-to-Actor Real-time Event Transport protocol (SARET) and Time-bound Event Transport Mode (TETM).

SARET protocol used an in-network congestion detection method, which is based on buffer occupancy of nodes, and an AIMD (Additive Increase Multiplicative

Decrease) based rate adjustment scheme for multi-event transport. The nodes assigned weight to individual event packets depending on their event priority, delay-bound and packet delay. In order to achieve deadlines, the packets with highest weights were forwarded first. Actors broadcasted reliability status to event nodes and the nodes adjusted their reporting rates in order to achieve required level of reliability.

The basic problem with SARET protocol was the rate adjustment scheme that used fixed increment and decrement factors. Selecting a small value of increment factor could take a lot of time to achieve reliability. Selecting a large value of increment factor immediately increases the reporting rate of nodes which could result into congestion on event occurrence.

TETM with destination guided reliability achievement mechanism used in-network based congestion mitigation and rate adjustment scheme. When destination guidance was decoupled from the rate adjustment mechanism then TETM achieved high system throughput. The proposed congestion control and schedule based rate adjustment method with little modifications were used. A delay constraint based packet forwarding scheme was introduced for forwarding in-time packets with shortest remaining delay bound first.

For actor to nodes information flow, an Actor to Sensor Reliable Transport (ASRT) protocol was presented in this study. ASRT was aimed to provide highly reliable information transport with minimum energy expenditure. Reliability in ASRT was defined as the successful delivery of all data packets from the source to all the sensor nodes. The basic design of ASRT protocol complemented the design of PSFQ (Wan, Campbell, & Krishnamurthy, 2005) protocol.

Like PSFQ, ASRT used in-sequence data forwarding and a NACK based data recovery mechanism. However, the operation of ASRT was different from PSFQ and according to the simulation results it was proved that the pump slowly and fetch quickly operation of PSFQ is only suitable for linear or sparse networks with low density. This study showed that, the rate at which errors (missing packets) should be recovered (NACKs sent) depends on arrangement of nodes in the network.

Nodes in sensor networks can be mobile; the degree of mobility depends on the nature of application. In the selected case study of mining application, the sensor nodes can be attached with the miners for precise position detection of a miner and for detecting survivors in disaster recovery situations. The transport of information from mobile nodes to actors is an important future direction for this study. In order to handle mobile nodes, the slot allocation mechanism of the proposed sensors-to-actors transport solution should be modified.

Time division based scheduling scheme's efficiency depend on the synchronization of nodes. With the increasing synchronization, the performance of the scheduling schemes will become better. The techniques presented by Elson, & Romer, (2003), Sichitiu, Elson, Estrin, & Shenker, (2003) and Younis, & Fahmy, (2005) can be used for time synchronization among nodes.

These techniques synchronize the nodes initially on network setup time and frequently update the synchronization of the network nodes. In the proposed sensor-to-actors transport, synchronization is only required on event occurrence. Therefore, new synchronization schemes are needed to be researched which can provide on-fly synchronization at event occurrence.

In case of actor to sensors transport, the ASRT protocol was designed to detect and recover lost packets. Bit errors or erroneous packets are not handled in ASRT protocol. Since actor to sensors transport in sensor networks is generally used to transport binary files, bit errors can not be ignored. As a future direction, the ASRT protocol can be enhanced to handle bit errors by using simple FEC based schemes.

ASRT and existing many-to-one transport solution used explicit NACK data recovery models. Since sensor networks use broadcast medium for information dissemination, implicit acknowledgement of received packets could be achieved by overhearing same packet transmissions from the neighboring nodes. Likewise, missing packet could be detected by overhearing the neighboring node transmission. This will decrease the overhead of NACK packet transmission and will decrease the energy consumption of nodes.

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